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Signalling message transport for B-ISDN

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Abstract

Signalling is the critical component of telecommunications networks. Thus the design of the signalling system is a fundamental issue for the creation of the future Broadband ISDN out of ATM switching technology. Today's telecommunication networks use common channel signalling networks (standardised in SS7) which are often separate from the bearer or transport network. With an ATM based Broadband ISDN (B-ISDN), it is anticipated that both signalling and user traffic will share the underlying ATM layer.

An SS7 "Message Transfer Part" (MTP) provides reliable connectionless signalling message transport. However due to differences between SS7 links and ATM signalling links (i.e. ATM Virtual Channel Connections (VCCs)), a new MTP will be needed for B-ISDN. This thesis is concerned with the design and engineering of the B-ISDN MTP.

The need for an ATM signalling link error monitor is identified as a key requirement. To design such an error monitor, the delay performance of ATM signalling links during errored periods must be determined. This requires a model for the delay performance of the Service Specific Connection Oriented Protocol (SSCOP), the link level protocol to be used on ATM signalling links. The thesis presents an SSCOP delay model, which also gives general results for mean Selective Repeat transmitter delay and the mean (and distribution) of Selective Repeat receiver delay.

Techniques for error monitoring on SS7 links are well established. The ATM layer and Signalling ATM Adaptation Layer also provide error monitoring capabilities. Hence there are several options for ATM signalling error monitors, based on either SS7 techniques or the new capabilities of the ATM protocols. The thesis compares

these options, and concludes that the most effective ATM error monitor is one that use the Operations and Maintenance (OAM) capabilities of the ATM layer. The implementation details of an OAM based error monitor are described.

In addition to ATM signalling link performance modelling and error monitoring, the thesis considers the wider issue of the functions required for the B-ISDN MTP. Central to these requirements is the signalling architecture which the B-ISDN MTP will support, i.e. fully meshed or STP based. These two architectures are compared, from the perspective of bandwidth requirements and maximum call handing capacity. The conclusion is that fully meshed ATM signalling networks are a viable alternative those which use STPs.

Accordingly, the thesis considers the structure of a B-ISDN MTP for a signalling network without STPs. This MTP is much simpler than its SS7 counterpart, due to the omission of the STP functionality. A "minimal" B-ISDN MTP is therefore proposed, which provides basic signalling message transfer capabilities with only a limited set of management capabilities. This minimal B-ISDN MTP would be suited to the needs of private networks, and complements the current B-ISDN MTP proposals (ie those in the ITU and the ATM Forum), which have increased levels of functionality appropriate to the needs of public networks.

The investigation of the current B-ISDN MTP proposals includes a critical examination of the proposed ATM signalling link proving scheme, which identifies a number of deficiencies. To overcome these deficiencies, a new ATM signalling link proving algorithm is introduced.

Acknowledgements

I would like to thank my supervisors, Professor Hugh Bradlow and Professor Gary Anido. It has been their help, encouragement and support which has made it possible to complete this work. I would also like to thank Telstra for their financial support , and the staff at Telecom Research Laboratories for their interest and help with this project.

I would also like to thank my colleagues; David Atkinson, Jeremy Lawrence, Eryk Dutkewitz, Lorraine De Vere and everyone else at the Switched Networks Research Centre. In particular I would like acknowledge the support given by Dr. Istvan Kerekes, and thank him for it. Professor Warren Yates, Dr. Aruna Seneviratne and Malcolm Butler also deserve thanks.

Finally I would like to thank my parents, Viv and Briar, for supporting me at all times, including the hard ones.

Table of Acronyms

AAL	ATM AdaptationLayer
AERM	Alignment Error Rate Monitor
AIS	Alarm Indication Signal
ARQ	Automatic Repeat Request
ATM	Asynchronous Transfer Mode
B-ICI	Broadband Intercarrier Interface
B-ISDN	Broadband Integrated Services Digital network
B-ISUP	Broadband Integrated Services User Part
BER	Bit Error Rate
BIB	Backward Indicator Bit
CBR	Constant Bit Rate
CCITT	International Telephone and Telegraph Consultative Committee (now the ITU-TS)
CLP	Cell Loss Priority
CRC	Cyclic Redundancy Test
CS1	Capability Set 1
EIM	Errored Interval Monitor
FERF	Far End Receiver Failure
FIB	Forward Indicator Bit
FISU	Fill In Signal Unit
GTT	Global Title Translation
HDLC	High-Level Data Link Control

IAM	Initial Address Message
IN	Intelligent Network
ISUP	Integrated Services User Part
ITU-TS	International Telecommunication Union Telecommunication Standardization Sector, (formerly the CCITT)
LSSU	Link Status Signal Unit
MID	Multiplexing Identifier
MSU	Message Signal Unit
MTP	Message Transfer Part
NNI	Network Network Interface
OAM	Operations and Maintenance
PCR	Preventive Cyclic Retransmission
PDH	Plesiochronous Digital Hierarchy
PDU	Protocol Data Unit
PVCC	Permanent Virtual Channel Connection
SAAL	Signalling AAL
SCCP	Signalling Connection Control Part
SDH	Synchronous Digital Hierarchy
SIO	Service Information Octet
SLS	Signalling Link Selection
SP	Signalling Point
SRSCT	Signalling Route Set Congestion Test
SS7	Signalling System number 7
SSCF	Service Specific Coordination Function
SSCOP	Service Specific Connection Oriented Protocol
STM	Synchronous Transfer Mode
STP	Signalling Transfer Points
SUERM	Signal Unit Error Rate Monitor
TCAP	Transaction Capabilities Application Part
TFC	Transfer Controlled
TUP	Telephone User Part
UNI	User Network Interface
UUI	User to User Information
VC	Virtual Channel
VCC	Virtual Channel Connection

VCI	Virtual Channel Identifier
VPC	Virtual Path Connection
VP	Virtual Path
VPI	Virtual Path Identifier
VSN	Virtual Service Network

Table of Symbols

Here we list the default meaning of symbols which are used frequently. Although an effort has been made to maintain consistent use of these symbols throughout the thesis, there are some exceptions (which are noted in the text).

λ	Arrival rate of new MSUs (or packets)
λ_e	Mean rate at which packets with errors on their first transmission arrive at the receiver
λ_p	POLL arrival rate.
λ_r	Arrival rate for retransmitted packets
ρ_m	Utilisation of transmitter queue by new MSUs (or packets) ($= \lambda S_m$)
ρ_r	Utilisation of the server by retransmitted packets
p_m	Message Signal Unit (MSU) error probability [1]
p_i	The error probability of a packet of length l_i ., where $p_i = 1 - (1-b)^{l_i}$
ρ_p	Utilisation of the server by POLL and STATUS packets

1. The terms MSU and packet are interchanged at times during this thesis. MSUs are referred to in the analyses which apply specifically to signalling messages (e.g. chapter 3). For the more general treatment of Selective Repeat delay in chapter 4, we refer to packets instead.

b	Bit error rate (independent bit errors are assumed)
$B(t)$	Customer service time distribution (used during general discussion of M/G/∞ queues).
D_a	Retransmission delay (i.e. from when the packet first leaves the server until a successful transmission is completed)
D_t	Total mean transmitter delay
m_i	Probability that an MSU has a length l_i bits $(1 \leq i \leq h)$.
Q_m	Mean packet queueing time, i.e between joining the queue and entering the server for the first time
S_e	Mean service time for errored packets
S_f	Fill In Signal Unit (FISU) service time (a constant)
S_i	The service time of a packet of length l_i
S_m S_{m2}	First two moments of MSU (or packet) service time (i.e. transmission time)
S_p , S_{p2}	First two moments of POLL service time.
S_r , S_{r2}	First two moments of the service time for retransmitted packets
S_s	Delay before a Selective Repeat transmission commences, due to the server being occupied by another packet.
S_v , S_{v2}	First two moments of the Virtual service time, i.e time for a packet to complete a successful transmission, once the initial transmission has begun.
T_d	Delay between an errored packet arriving at the receiver and the error being detected by the SSCOP protocol.
T_L	Round trip time, i.e. twice the propagation delay plus processing time at the receiver.
u	Packet service rate (packets/sec)

Table of Contents

1. Introduction

1.1 Background

The SS7 signalling system is a key component of the telecommunications network. SS7, or CCS7, has been evolving over the last fifteen years. While originally introduced to support basic telephony, SS7 also provides the signalling infrastructure needed for ISDN, Mobile and Intelligent Network Services.

SS7 traffic is carried by a dedicated packet switched network, which has a rigidly defined architecture, based on 64 kbps links (56kbps in the US). Due to the critical role of signalling, SS7 protocols are designed to ensure continued operation during signalling network failures and overloads. ITU recommendations Q.700 to Q.710 define an SS7 "Message Transfer Part" (MTP), which provides reliable connectionless signalling message transfer for all signalling applications. The MTP corresponds roughly to the bottom 3 layers of the OSI reference model.

The advent of B-ISDN, and the use of the Asynchronous Transfer Mode (ATM) switching technique, will change significantly the way signalling traffic is carried. ATM divides all traffic into fixed length, self routing cells, which are directed through the network by fast hardware switches. The ATM system provides a logical network of Virtual Channel Connections (VCCs).

A fundamental difference between ATM and Synchronous Time Division Multiplexing (the transport system for SS7) is that the bandwidth of an ATM VCC is not fixed to a basic rate (or multiple thereof). Indeed the bandwidth of an ATM VCC may change during the lifetime of a connection. Also ATMs VCCs may share bandwidth amongst each other.

The B-ISDN network will use ATM VCCs instead of SS7 links to carry signalling traffic. Hence the SS7 MTP will need to be modified substantially for B-ISDN, to account for the different characteristics of these ATM signalling[1] links. An important part of this re-engineering process will be the development of new error monitoring methods, which in turn will require new techniques for delay performance modelling. In addition, the low cost of ATM VCCs (compared to current SS7 links) makes possible a new signalling network architecture, with few (or no) Signal Transfer Points (STPs). This new ATM signalling architecture, its accompanying Message Transfer Part, and the performance modelling and error monitoring of the component ATM signalling VCCs are the topics of this thesis.

1.2 Overview

The thesis has 7 chapters. They are organised as follows.

Chapter 2 reviews the current SS7 network and protocols, focusing on the MTP. The major characteristics of B-ISDN, and the options for ATM based signalling transport are also outlined. This review has two aims: to provide the technical background for the thesis and to identify the key engineering issues for the B-ISDN MTP. These issues are addressed in the remainder of the thesis.

A major requirement for the B-ISDN MTP will be an error monitor for the component ATM signalling links (or VCCs), which indicates when link per-

1. The spelling conventions used throughout this thesis are those of the Macquarie Dictionary, Macquarie University, Australia, 1991

formance has become degraded (so that a link changeover may be done). To design this error monitor, the delay performance of ATM signalling links must be modelled (as the link delay associated with a given error rate forms the basis of a decision to changeover the link). To this end, chapter 3 reviews previous analyses of SS7 signalling link delay. In addition, previous Selective Repeat delay analyses are reviewed, as Selective Repeat ARQ has been chosen for ATM signalling (SS7 uses go-back-n). The deficiencies identified with these analyses indicate a clear need for a new Selective Repeat performance model. This new model is presented in chapter 4.

Chapter 3 also reviews previous studies of SS7 error monitors. These studies provide the criteria for evaluating ATM error monitor effectiveness. We identify three potential ATM error monitors, two of which are currently used on SS7 links. The third uses the ATM VCC Operation and Maintenance (OAM) capabilities. A comparative study of these three error monitors is done in chapter 5.

Chapter 4 investigates ATM signalling delay performance. This begins with an outline of the Service Specific Connection Oriented Protocol (SSCOP), the link layer protocol developed for the Signalling ATM Adaptation Layer (AAL). Then a new ATM signalling (i.e. SSCOP) delay performance model is presented. This gives mean delay and delay percentiles for the SSCOP (and more generally for the Selective Repeat ARQ), while addressing the deficiencies with previous Selective Repeat analyses noted in chapter 3. The SSCOP performance results form the input to the error monitor design described in chapter 5.

Chapter 5 considers error monitors for ATM signalling links. We begin by determining maximum sustainable error rates for ATM signalling links, based on the SSCOP results from chapter 4. This leads to a comparison of the three potential error monitoring schemes for ATM signalling. By considering the error monitor performance criteria described in chapter 3, as well as the Signalling AAL operation, we conclude that an OAM based error monitor is the most suitable choice for ATM signalling links.

Chapter 6 moves from ATM signalling link performance modelling and error monitoring to the wider issue of the functions required for the B-ISDN MTP. Central to these requirements is the signalling architecture which the B-ISDN MTP will support, i.e. fully meshed or STP based. We present a case study which compares these two architectures, from the perspective of bandwidth requirements and maximum call handing capacity. The conclusion is that fully meshed ATM signalling networks are a viable alternative to those which use STPs.

The remainder of chapter 6 examines, accordingly, the structure of a B-ISDN MTP for a signalling network without STPs. Recognising that such an MTP can be much simpler than its SS7 counterpart, we propose a "minimal" MTP which provides basic signalling message transfer capabilities with only a limited set of management capabilities. This minimal B-ISDN MTP would be suited to the needs of private networks, and complements the more complex B-ISDN MTP proposals of [ATM94.1], [ATM94.2] and [ITU94.5], which provide a level of functionality more appropriate to the needs of public networks.

The investigation of the B-ISDN MTP includes a critical examination of the proposed ATM signalling link proving scheme proposed in [ITU94.5]. A number of deficiencies are identified. To overcome these deficiencies, a new ATM signalling link proving algorithm is introduced.

This thesis examines key aspects of the B-ISDN MTP. Arising from this examination are a series of proposals and recommendations for the operation of the B-ISDN MTP, at both the link level and the network level. These proposals and recommendations are summarised in chapter 7, as well as further areas of research.

1.3 Contributions

This section lists the original contributions contained in this thesis, along with the section where the relevant work is first discussed.

1) Developed a new analytic expression for the mean Selective Repeat transmitter delay, which (unlike previous analyses) accounts for variations in packet size and round trip delay (Chapter 4, section 3).

2) Developed an new analytic expression for the Selective Repeat receiver delay distribution, which (unlike previous analyses) accounts for variations in packet size and round trip delay. This method provides a simple closed form expression for the mean Selective Repeat receiver delay (Chapter 4, section 3).

3) Adapted the above methods to provide a delay performance model for the Service Specific Connection Oriented Protocol (SSCOP), the link level protocol to be used for B-ISDN signalling (Chapter 4, section 3).

4) Used the SSCOP performance modelling results to determine maximum sustainable error rates for ATM signalling links (Chapter 5, section 3)

5) Developed a new method for calculating the mean time to overflow an SS7 Leaky Bucket. This also provides the first passage time (i.e. time for the first overflow) of an M/G/1/K queue (Appendix 1).

6) Designed an ATM signalling link error monitor (Chapter 5, sections 5 and 7).

7) Demonstrated the feasibility of a fully meshed B-ISDN signalling network, compared to an STP based alternative, in terms of bandwidth requirements and call handing capacity (Chapter 6, section 3).

8) Designed a minimal Message Transfer Part (MTP) for a B-ISDN Signalling Network which uses Associated Mode signalling only (i.e. without STPs) (Chapter 6, section 4).

9) Designed an ATM signalling link proving scheme (Chapter 6, section 5)

1.4 Publications

The work for this thesis has resulted in the following publications.

1. Bradlow H., Eyers T[2]., "Delay Performance of B-ISDN Signalling Links", Australian Telecommunications Research, Vol. 25, No. 1, 1991, pp 37-44

2. Eyers T., Bradlow H., "Performance of ATM Signalling Links", ABSSS, Sydney, July, 1991

3. Eyers T., Bradlow H., Anido G., "ARQ Protocols for B-ISDN Signalling Message Transfer", 6th Australian Teletraffic Research Seminar, Wollongong, November, 1991

4. CCITT Study Group XI, Contribution no. D.2075, Australia, March, 1992

5. Eyers T., Bradlow H., Anido G., "Monitoring Error Performance of B-ISDN Signalling Links", ABSSS, Melbourne, July, 1992

6. Eyers T., Bradlow H., Anido G., "Practical Selective Repeat ARQ for B-ISDN", 7th Australian Teletraffic Research Seminar, River Murray, South Australia, November, 1992

7. Kerekes I, Anido G, Bradlow H., Eyers T., "Dimensioning of Leaky Buckets to be Used as Monitors", Electronics Letters, Vol. 29, No. 2, January 1993, pp 245-246

8. Eyers T., Bradlow H., Anido G., "Signalling Message Transfer Part for B-ISDN", ABSSS, Wollongong, July, 1993

9. Eyers T., Bradlow H., Anido G., "B-ISDN Signalling Architecture", 8th Australian Teletraffic Research Seminar, Melbourne, December, 1993

2. The publications of the author are shown as "Eyers T." (T. standing for Tony), instead of the formally given names "Anthony William", which appear at the front of the thesis.

10. T. Eyers, H.S. Bradlow, G. Anido, "Selective Repeat Receiver Delay with Application to the SSCOP", Electronics Letters, Vol. 30, No. 19, September 1994, pp1579-1581

11. T. Eyers, H.S. Bradlow, G. Anido, "Performance Monitoring for ATM Signalling Link Changeover", Australian Telecommunications and Applications Conference, Melbourne, December 1994

2. SS7 and B-ISDN Signalling Overview

2.1 Introduction

This chapter reviews the SS7 and B-ISDN architecture and protocols, providing the technical background needed for this thesis. This review also identifies the engineering issues arising from a B-ISDN MTP implementation. These issues are addressed in subsequent chapters.

Section 2.2 begins with an overview of the SS7 MTP, which provides signalling applications with a service similar to the first three layers of the OSI reference model [TAN88]. As this thesis aims to guide implementations of the B-ISDN MTP, a clear understanding of the SS7 MTP is a key requirement. The standard SS7 network architecture is presented in section 2.3, followed in section 2.4 by descriptions of the key SS7 MTP procedures relating to this work; namely Error Monitoring and link Changeover. The suitability of the SS7 Error Monitoring procedures for ATM signalling links is subsequently examined in chapter 5. SS7 congestion controls are reviewed in section 2.5.

Section 2.6 introduces B-ISDN, ATM switching techniques, and the protocols that will interface signalling applications to the ATM network. Section 2.7 outlines alternatives for B-ISDN signalling transport. The purpose of these sections is to show the difference between using 64 kbps circuits to carry signalling (the SS7 approach) and using ATM VCCs. Of particular importance is the performance monitoring capabilities provided by the ATM layer. These capabilities, which are outlined in this chapter, form the basis of the ATM signalling Error Monitor proposed in chapter 5.

Drawing on the background given in the previous sections, section 2.8 identifies key engineering issues for the B-ISDN MTP. These issues, namely error monitoring and performance modelling of ATM signalling VCCs, B-ISDN signalling architecture, and determining MTP protocol functions required for B-ISDN, are addressed in the remainder of the thesis.

2.2 SS7 Signalling Transport

The first set of SS7 standards were published in 1980 by the CCITT (now known as the ITU-TS). Known as the "yellow book" standards, they were revised in 1984, 1988 and 1992. A key task of these standards was to define common procedures for signalling message transport, with the aim of providing a reliable connectionless datagram service (i.e. the SS7 MTP). While the initial task for the MTP was to transport the signalling for telephony, the modular design of SS7 allows MTP to support all signalling applications.

Here we give a general overview of the MTP. Subsequent sections then describe in more detail the MTP procedures which are relevant to the work in this thesis.

2.2.1 Message Transfer Part

The MTP is specified in [Q.700], with tutorial overviews in [MOD90], [LAW88] and [JAB91]. Figure 2.1 shows the MTP structure, and its relationship to the OSI reference model. An additional component, known as the Signalling Connection Control Part (SCCP), provides connection establishment capabilities and expanded addressing. MTP and the SCCP combine to

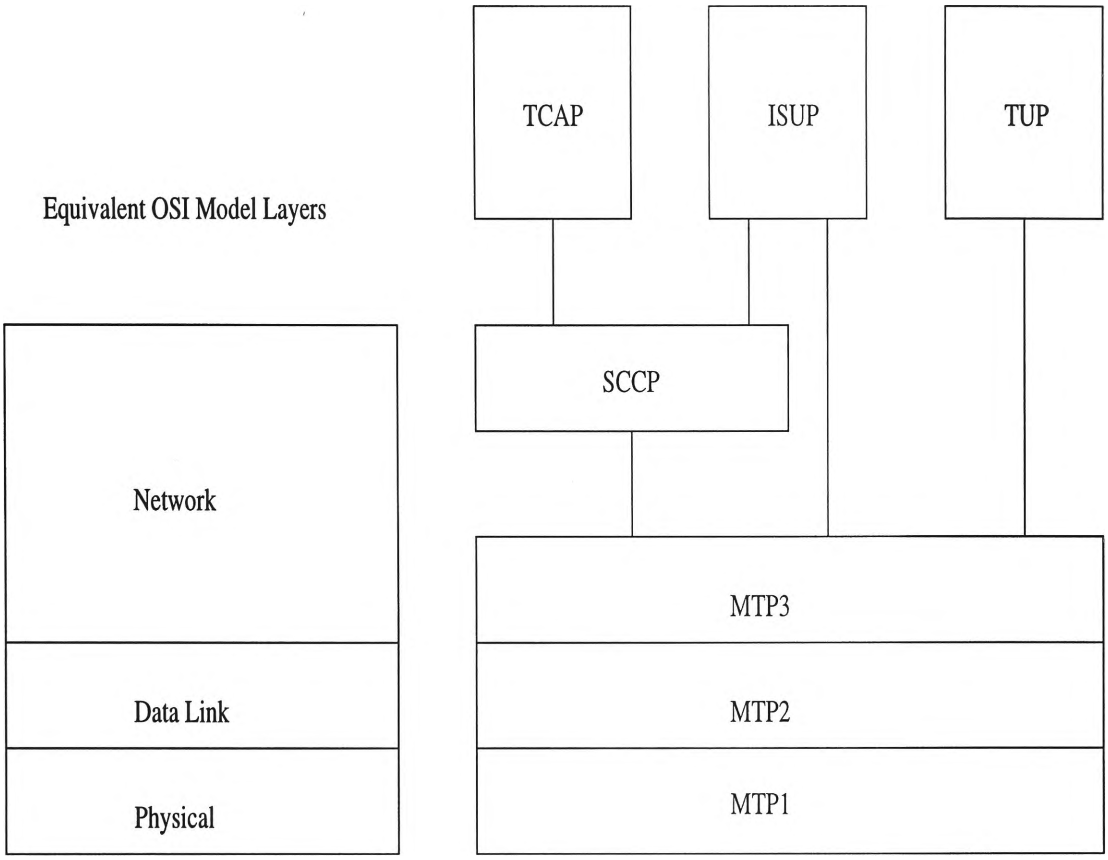


Figure 2.1 SS7 Protocol Stack

form the Network Services Part, which provides a similar service to the first three OSI layers. Minor differences are described in [MIT91].

The MTP (and in some cases the MTP plus the SCCP) provide a common platform to transport messages from SS7 User Parts. Figure 2.1 shows three such User Parts, the Telephone User Part (TUP), the ISDN User Part (ISUP) and the Transaction Capabilities Application Part (TCAP). TUP and ISUP do the network level signalling to establish telephone and ISDN calls respectively (while both User Parts are shown here, an SP would normally have either ISUP or TUP, not both). TCAP allows an application at one node to invoke procedures at another node, and subsequently exchange the results. (for example database lookups for Intelligent Network services). The combination of MTP, SCCP and User Parts provide the SS7 signalling capability.

The structure shown in Figure 2.1 shows the layers which combine to form the MTP. A brief description of each layer follows.

2.2.2 MTP1

The MTP physical layer, known as MTP1, provides a bi-directional 64 kbps transmission path (56 kbps in the US). Lower data rates are allowed, the minimum being 4.8 kbps, while high speed (1.536 Mbps) links are planned for US networks [SCH94]. To increase the signalling capacity between two adjacent nodes (known as Signalling Points (SPs)), multiple signalling links operate in parallel. A collection of signalling links with a common source/destination is known as a signalling link set [Q.701]. A signalling link set has a maximum of 16 links (32 links in the US).

2.2.3 MTP2

MTP2 provides a link level protocol similar to High-level Data Link Control (HDLC) [STA89], but with different link establishment procedures and dedicated link monitoring capabilities. The operation of MTP2 is defined in [Q.702]. Here we describe briefly the MTP2 functions, beginning with error correction.

MTP2 error correction operates in two modes, Basic and Preventative Cyclic Retransmission. Basic mode, used for terrestial links, employs the go-back-n error correction (or Automatic Repeat Request (ARQ)) technique [STA89]. While the go-back-n method is easily implemented, it gives poor delay performance at high error rates (compared to the Selective Repeat ARQ [EYE91]). The relatively poor performance of go-back-n is due to its retransmission of all unacknowledged packets when an error is detected (Selective Repeat retransmits the errored packet only). This disparity in performance becomes more pronounced as the link bit rate increases, due to a higher number of unacknowledged packets (for a given utilisation). The higher link bit rates expected in ATM networks (compared to SS7) hence justify the choice of Selective Repeat ARQ for B-ISDN signalling, as will be discussed in chapters 3 and 4.

Preventive Cyclic Retransmission (PCR) is used on satellite links. Here the extra delay (compared to terrestial links) for the arrival of negative acknowledgements means that a go-back-n protocol would cause large delays during error periods. PCR avoids this problem by retransmitting unacknowledged MSUs when there is no new traffic to send, without waiting for negative acknowledgements. The drawback of PCR is that it uses bandwidth less efficiently than go-back-n. PCR performance is studied in [SKO88].

As SS7 message transfer delays add directly to call setup delays, an important function of MTP2 is to limit the delay increase due to errors (and the resulting retransmissions). This is done by removing a signalling link from service once its error rate exceeds a prescribed limit. This error monitoring is done with a Leaky Bucket counter (called a Signal Unit Error Rate Monitor (SUERM) in [Q.703]). In addition to detecting high error rates, the Leaky Bucket error monitor detects link outages (when no Signal Units arrive). Once a link has been failed (i.e. removed from operation because of degraded performance), MTP2 initiates an automatic test procedure, known as link alignment, to determine when the link may carry live traffic again.

To ensure rapid detection of faults, SS7 links transmit 5 octet Fill In Signal Units (FISUs) continuously in the absence of other signalling traffic. FISU format is shown in Figure 2.2. While providing MSU acknowledgement, and activating the Leaky Bucket counter, errored FISUs are not retransmitted. By sending these FISUs during idle periods, SS7 links always maintain a constant bit rate, regardless of the offered load. However, this constant bit rate will be not maintained on ATM VCCs, as they will not use FISUs [ITU94.5]. One reason for this is that the acknowledgement of MSU arrivals, an important task of the FISU on SS7 links, is achieved by other means on ATM signalling links. The implications of a variable rather than a constant bit rate for ATM signalling error monitoring will be addressed in chapter 5.

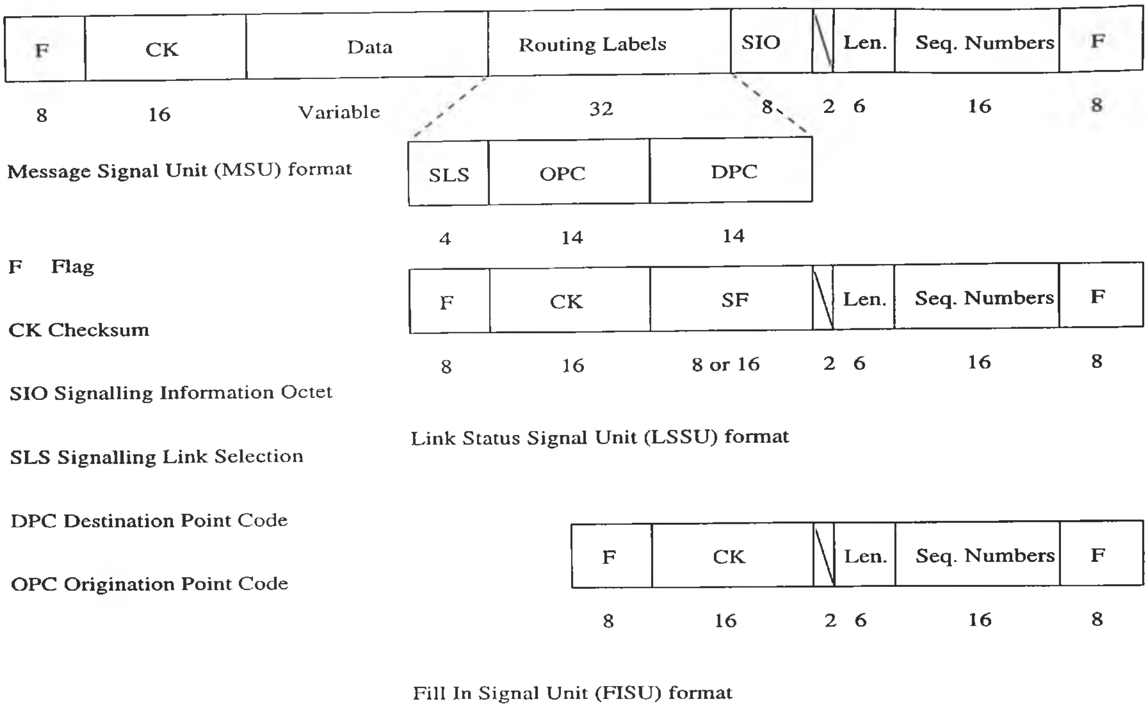


Figure 2.2 Signal Unit Formats

MTP2 flow control is done by periodically sending management messages, called Link Status Signal Units (LSSUs), which indicate that the link is "Busy". LSSU format is shown in Figure 2.2. Acknowledgements (both positive and negative) are withheld while the congestion is pending. If congestion lasts for longer than 3-6 seconds, the link is failed.

Processor Outage refers to a failure above OSI level 2 (e.g. level 3). Such a failure is advertised by periodically sending LSSUs with "Signalling Indication Processor Outage". The remote end, upon receiving such LSSUs, informs Level 3, so that signalling traffic may be diverted.

2.2.4 MTP3

MTP3 provides a connectionless datagram service for the SCCP and the user parts. In addition, MTP3 recovers from network level problems, due to link failure or congestion. MTP3 functions fall into two major categories; signalling message handling and signalling network management [Q.704].

Signalling message handling comprises message routing, discrimination and

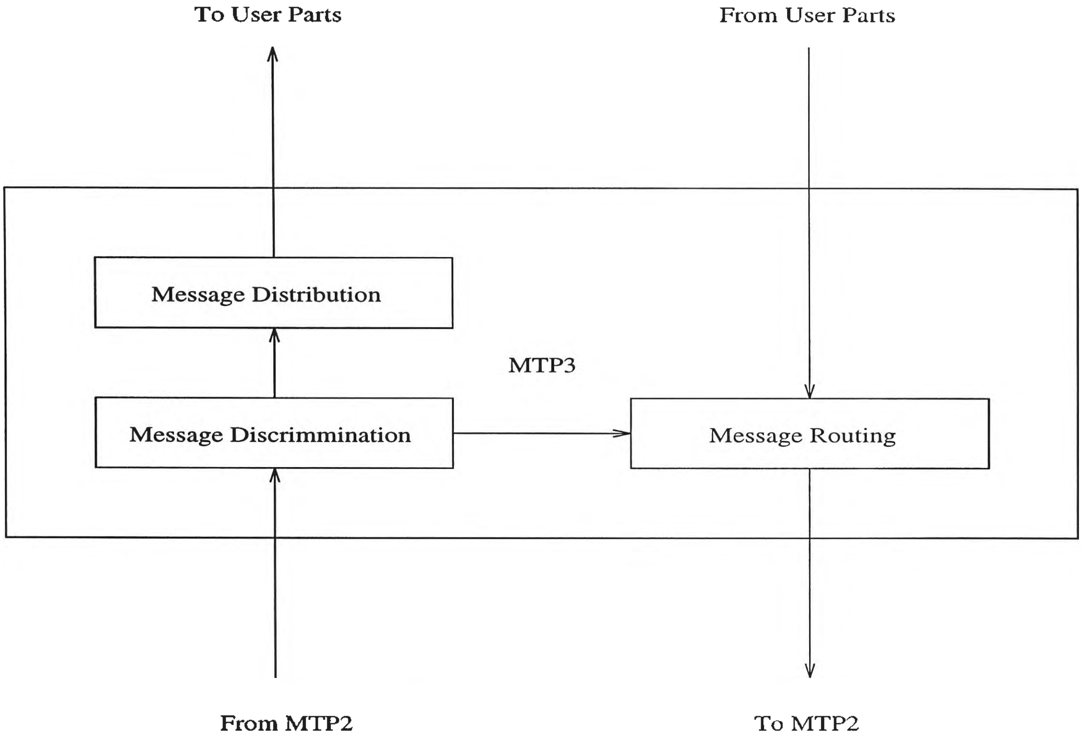


Figure 2.3 MTP3 Structure

distribution, as shown in Figure 2.3. Message discrimination decides if a Message Signal Unit (MSU) has reached its destination, or whether it is to be routed to another SP, or Signal Transfer Point (STP). STPs are essentially packet switches which route signalling messages, thereby acting as transit nodes. Message discrimination is based on the 14 bit Destination Point Code (24 bits in the US) carried in each MSU (as shown in Figure 2.2). If an MSU has not yet reached its destination, it is passed to the message routing process, which determines the outgoing link. Otherwise the MSU is passed to the message distribution process, which determines the destination User Part, based on a 4 bit field within the Service Information Octet (SIO). The SIO is part of the MSU header, shown in Figure 2.2.

To maintain an even load on the links within a signalling link set, the "SLS" algorithm is used. Each MSU header carries a 4 bit (5 bit in the US) Signalling Link Selection (SLS) field, whose value determines which link (from a given link set) will carry the MSU. An even load is maintained within a linkset by rotating the SLS value amongst the MSUs. All MSUs for a given call are given the same SLS value, which ensures that they follow the same route, and hence arrive in sequence (usually connectionless networks, such as IP, cannot guarantee sequencing).

The operation of the SLS scheme, and the change in load distribution as the linkset size changes, is described in [WAN91]. A drawback of the scheme is that it places an upper limit on the number of links (and hence the bandwidth) in a linkset. In particular, this limits the signalling capacity between Signal Transfer Points (STPs). To increase this capacity, additional STPs must be deployed [CAM92]. This is a major shortcoming of SS7, as network providers are obliged under these circumstances to deploy extra STPs, just to obtain extra bandwidth. This restriction on bandwidth however should not apply in an ATM signalling network, as the capacity of individual VCCs is not likely to be restricted to a single value (the case with SS7 links).

The purpose of Signalling Network Management is to reconfigure the network in the event of failure or congestion, to avoid message loss/duplication and minimise delay. This process has 3 major components; traffic management, route management and link management.

Signalling Traffic Management ensures that link changeovers and changebacks occur without message loss or duplication. Similarly, forced rerouting and controlled rerouting procedures divert traffic when signalling routes (one or more signalling links in tandem) become unavailable or available, respectively. A Signalling Point Restart procedure is provided, which ensures that an SP has received all information regarding unavailable links/routes before it starts sending signalling messages.

The purpose of Signalling Route Management is to distribute information regarding congested and unavailable routes. Signalling Link Management procedures are used to activate and deactivate signalling links. In particular, they initiate the MTP2 link alignment procedure, which must complete successfully before a link carries live traffic.

2.2.5 SCCP

The SCCP enhances the services of MTP in two ways; by allowing different classes of connection service and by providing extended addressing.

SCCP offers 4 connection classes. These are:

Class 0: Basic Connectionless Class

Class 1: Sequenced Connectionless Class

Class 2: Basic Connection Oriented Class

Class 3: Flow Controlled Connection Oriented

Class 0 is the basic MTP connectionless service. Class 1, while still connectionless, ensures that messages arrive in sequence (this may be achieved by ensuring that all messages from a particular sequence have the same value in the SLS field). Class 2 provides a basic connection oriented service, with a segmentation and reassembly capability to allows the transmission of long messages (MTP2 imposes a maximum Signal Unit length of 272 octets). Class 3, in addition to providing flow control, uses sequence numbers to detect message loss. If this occurs, the signalling connection is reset and notification given to the higher layers.

SCCP extends the addressing capability of MTP3 by providing a Subsystem Number Field, which allows up to 256 separate users to be identified at a destination SP (MTP3 addressing allows only 16 separate users). In addition, SCCP allows signalling messages to be routed using a variety of addressing schemes (e.g. e.164, x.121). The procedure for this is called Global Title Translation (GTT). GTT inputs a called address, and converts it to a Destination Point Code and Subsystem Number, so that the message may be routed

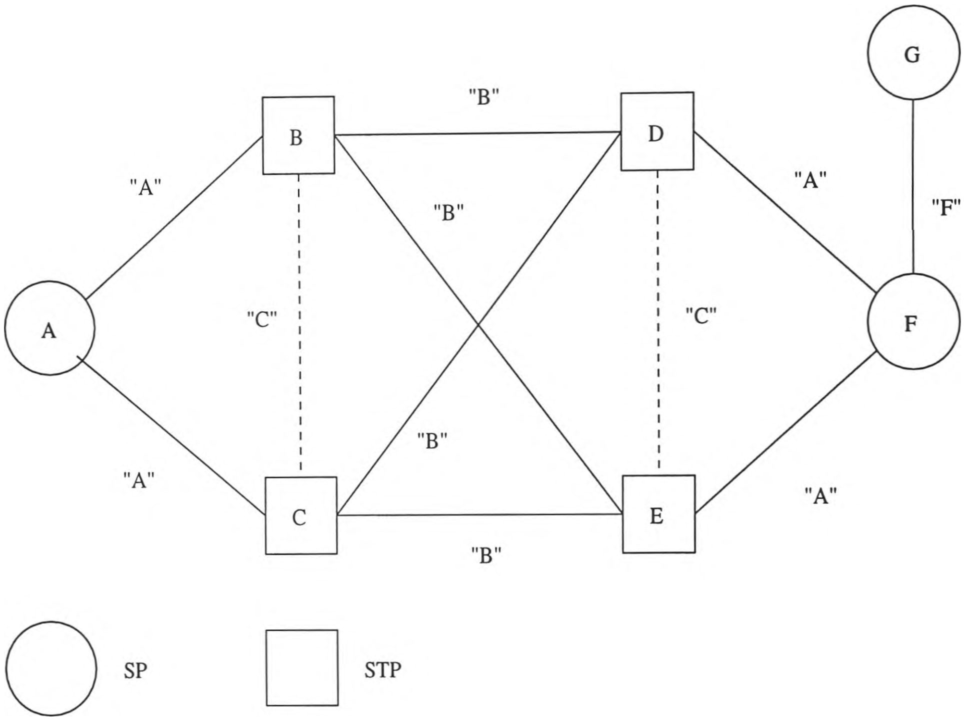


Figure 2.4 SS7 Quad Architecture

by MTP3. This capability may used to allow SPs to reach network databases without needing to know their Destination Point Code (for example, when 800 number translations are done). [CLA88] shows an example where a designated STP uses GTT to direct messages from a number of SPs to an given database. The actual location of the database is transparent to the individual SPs, and need only be known to a single STP.

2.3 SS7 Network Architecture

This section reviews the standard SS7 network architecture and the alternatives for SS7 message transfer.

ITU-TS recommendation Q.705 specifies a mean unavailability of 10 minutes per year for a signalling route set (i.e. all the signalling routes between 2 SPs). To achieve this unavailability figure, redundancy is designed into SS7 networks. A common method for achieving this redundancy is the "quad" architecture shown in Figure 2.4 [Q.705] [MOD90]. STPs are deployed in mated pairs, so that if one fails the other will take its load. Two such pairs are shown in Figure 2.4, B/C and D/E. During normal operation (i.e. no failures), 4 routes between SPs A and F are used; A-B-D-F, A-B-E-F, A-C-E-F and A-C-D-F. The SLS field, described in the previous section, is used to share traffic evenly among the 4 routes [Q.705].

Figure 2.4 shows 4 standard link types: "A" links connecting SPs to their local STP pair, "B" links interconnecting STP pairs, "C" links (the dotted lines), which go between the STPs in a mated pair and "F" links, which go directly between SPs. "C" links are used only after link failures. For example, after a failure on the D-F link, traffic arriving at STP D (for SP F) is diverted to STP E, via the "C" link.

The architecture shown in Figure 2.4 is commonly used in US networks. Other paradigms exist, for example the Japanese Signalling Network described in [KIT90]. Here the SPs are divided into twelve regions, with an STP pair serving each region. These STPs are divided into two planes, A and B, with each STP in a pair belonging to a different plane. The twelve STPs in each plane are fully meshed.

The STP based architecture gives rise to two signalling "modes", namely the associated and quasi associated mode [Q.700]. Signalling messages using the associated mode travel directly between source and destination SPs (usually on the same path as the bearer connection), without passing through STPs. Messages using the quasi-associated mode traverse one or more STPs.

The quad architecture in Figure 2.4 employs the quasi-associated mode. For associated mode signalling, direct links are needed between SP pairs. These direct links, known as "F" links (e.g. between SPs F and G), are deployed only if there is sufficient demand between an SP pair to justify the cost of the extra link [STA87].

2.4 SS7 Error Monitoring

The previous two sections present a broad overview of SS7 message transport. A critical function of SS7 is to monitor link error performance, and to enforce a link changeover when the error performance falls below a specified limit. This is to ensure that the message transfer delay remains within specified bounds (i.e. those specified in ITU-TS recommendation E.733) This section describes the SS7 MTP procedures of Error Monitoring, Link Changeover and Link Alignment. The key task of extending these SS7 procedures to B-ISDN signalling is addressed in subsequent chapters.

MTP2 employs two error monitoring procedures. The Signal Unit Error Rate Monitor (SUERM) measures the error rate on a working link, to determine if it should be taken out of service. The Alignment Error Rate Monitor (AERM) determines whether an out of service link is performing sufficiently well to carry live traffic.

The SUERM is incremented for each Signal Unit that fails the MTP2 acceptance procedure (i.e. for each error). Signal Units are not accepted if they

- are less than 6 octets or more than 272 octets

- do not contain an integral number of octets

- have a CRC error

The SUERM is decremented (provided that it is not 0) for every D Signal Units (D = 256), regardless of whether they are errored or not. A link failure is indicated if the SUERM reaches a level T (T = 64). If alignment is lost, indicated by more than 7 "1s" in a row, then the SUERM enters "octet counting" mode. Here the SUERM is incremented for every N octets received (N = 16), until a correct Signal Unit arrives, at which time the octet counting mode ceases (and normal error counting mode of the SUERM resumes). Essentially the octet counting mode equates loss of alignment for an N octet interval with a single Signal Unit error (i.e. both events result in an increment of the SUERM). The SUERM parameter values mean that a link failure will be declared if alignment is lost for more than 128 msec (i.e. the time for 64×16 octets to pass).

SS7 Link Alignment is a procedure which must complete successfully before a new link may be brought into service, or before a failed link may be returned to service. The purpose of this procedure is to ensure that the error rate is below a specified value before the link carries live traffic. The Link Alignment procedure invokes the Alignment Error Rate Monitor (AERM), which counts Signal Unit errors during a fixed proving period. As soon as 4 errors are counted during a Normal proving period (2^{16} octets or 8 seconds), or 1 error during an Emergency proving period (2^{12} octets or 0.5 seconds), then the proving period is aborted, and a new one begins immediately (conversely a successful link alignment is recorded if the error count is less than this, following which the link enters service). If proving is aborted M times (M = 5), then the Link Alignment procedure ceases, and the new (or previously failed) link is taken out of service[1].

The key function of the SUERM is to initiate a link changeover when the MSU delay (as measured (indirectly) by the error rate) exceeds a set limit. An additional function of the SUERM is to limit the changeover transient (i.e. the amount of data queued when link changeover occurs). However, as shown in [SCH94], the SUERM does not achieve this when the link speed is raised from 64 kbps to 1.536 Mbps (the reasons for this are outlined in chapter 3). To overcome this deficiency of the SUERM at high error rates, a new error monitor, which counts errored time intervals instead of Signal Units (and is hence known as the Errored Interval Monitor (EIM)), will be used in the US for 1.536 Mbps signalling links [SCH94]. The EIM increases a counter by U if an interval contains one or more Signal Unit errors, and decrements the counter by D for each error free interval. A changeover occurs

1. The ITU SS7 standard describing this procedure (Q.703) does not indicate what happens after a link is taken out of service. For SS7 links in the USA, a new link alignment procedure begins immediately in this instance [RAM93]. If Link Alignment does not complete within 480 to 600 seconds, then further Link Alignment attempts are halted, and manual intervention is sought.

when the counter reaches T (the proposed values for U, D and T are 144292, 9308 and 577169, respectively, with the interval being 100 msec).

2.4.1 Changeover Procedures

After a link failure has been declared, a changeover to an alternate link occurs. Similarly, a changeback is done when a previously failed link passes the Link Alignment procedure. A major aim of these procedures is to minimise message loss and duplication during changeover/changeback. We begin by describing link changeovers.

MTP3 initiates a link changeover when MTP2 indicates a failed link. The criteria for link failure are:

- excessive error rate or loss of alignment (indicated by the SUERM or EIM)
- excessive delay for MSU acknowledgements (0.5-2.0 sec)
- failure of signalling terminal equipment
- two out of three back sequence numbers (i.e. MSU acknowledgements) which do not correspond to MSUs in the retransmission buffer
- reception of consecutive LSSUs indicating link out of service or out of alignment
- excessive periods (3-6 sec) of Level 2 congestion at the remote end of the link (this is flagged by the arrival of an LSSU with a status indication of "BUSY").

When a link is recognised to be unavailable, MTP3 sends a management MSU (called a changeover order) to the remote end. This changeover order has the sequence number of the last MSU accepted by MTP3. MSUs arriving for the unavailable link after the changeover order are buffered. Upon receiving the changeover order, the remote end sends a changeover acknowledgement, which also has the sequence number of the last MSU accepted. This exchange of sequence numbers allows each end to then retrieve from MTP2 the MSUs that have not arrived at the other end. These retrieved MSUs are placed on to an alternate link, followed by the MSUs buffered after the changeover messages were sent. This handshake procedure ensures that a) no messages are lost or duplicated due to the changeover and b) both ends changeover at the same time.

If a signalling terminal cannot determine the sequence number of the last MSU accepted, then an "emergency" changeover is done. This involves the exchange of emergency (as opposed to normal) changeover messages. No message retrieval is done during an emergency changeover.

If changeover messages cannot be exchanged, due to a complete link failure or a processor outage, then a time controlled changeover is done. Traffic arriving for the unavailable link is buffered for 500-1200 msec, then a changeover to an alternate link is done (in this instance, unacknowledged messages are lost). The purpose of this delay is to avoid out of sequence messages.

Changeovers may also be done at the route level (a signalling route is one or more signalling links in tandem). Here the SP learns of the failed route by receiving an MTP3 "Transfer Prohibited" message, which comes from an STP which can no longer send messages to a particular destination. Upon receiving this message, the SP buffers traffic for the failed route, finds an alternate route (if possible), then diverts the buffered traffic to this new route. This procedure is called "forced re-routing" [Q.704]. Here message loss may occur, as the affected STP will have to discard the messages which it can no longer route.

The MTP3 link changeback procedure is used when a previously unavailable link returns to service, after completing MTP2 proving. The aim is to divert traffic from the alternate link back to the original (previously unavailable)

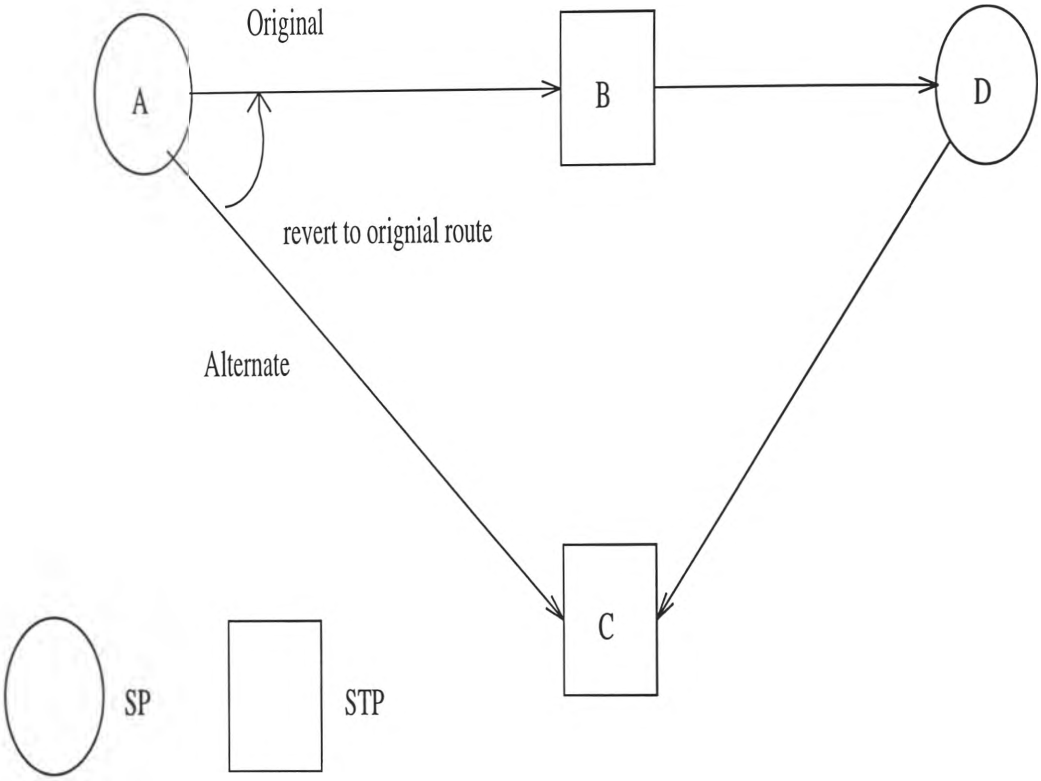


Figure 2.5 Time Controlled Diversion

link, without losing message sequence integrity. The procedure is to send an MTP3 "Changeback Declaration" on the alternate link, then buffer any traffic arriving for the original link (i.e. the one which is about to be restored). Upon receiving the Changeback Declaration, the remote end responds with an MTP3 "Changeback Acknowledgement" message. When this arrives, traffic is restored to the original link, starting with the buffered messages. Essentially the changeback procedure enforces a delay of one round trip (i.e. the time to exchange these two messages) before diverting any traffic. This ensures that any new messages sent on the restored link do not arrive out of sequence (i.e. before any related messages sent previously on the alternate link).

The link changeback procedure is used when both the original and alternate links have the same endpoint. If they have different endpoints, as shown in Figure 2.5, then a "time controlled diversion" is used to restore traffic to the original link. This has the same aim as the changeback procedure, i.e. avoiding out of sequence messages. This is done by enforcing a delay (500-1200 msec) between the last message on the alternate link and the first message on the restored original link. The time controlled diversion is also used when a failed route (initially indicated by an MTP3 "Transfer Prohibited" message) becomes available once more.

2.5 Congestion Control

SS7 error monitoring and changeover procedures are designed to recover from link/route failures. SS7 congestion control procedures aim to maintain signalling network operation during overload (often caused by link/route failures [MAN93.1]). Three schemes are specified, one for International signalling networks, the other two for National networks. These schemes are outlined here.

All three schemes detect congestion onset by monitoring the transmission queue length on each link (in addition, congestion controls may be invoked when the level 3 (i.e. MTP3) queue exceeds set thresholds [RUM94.1]). We consider first the International scheme, which maintains a single onset and abatement threshold (note: the "International" congestion control scheme is used in some European national networks, e.g. Germany [ZEP94]). When the onset threshold is exceeded, MTP3 "Transfer Controlled" (TFC) messages are returned to SPs sending traffic to the congested link, at a rate of 1 per 8 MSUs. TFCs continue to be sent until the queue drops below the abatement threshold. When a TFC arrives at an SP (or STP), MTP3 marks the indicated destination as congested, and sends congestion indications to each User Part trying to access it. The User Parts then reduce the rate of traffic to the congested destination. It is assumed that congestion has abated when TFC messages no longer arrive (i.e. there are no explicit procedures to indicate congestion abatement).

Two congestion control schemes are defined for National networks, one with and the other without MSU priorities. We begin by describing the priority scheme, where MSUs are assigned one of 4 priorities (i.e 0-3), according to their function. The priority assigned to each MSU is indicated by two bits in the Service Information Octet (SIO), which forms part of the MSU header. Each priority level (apart from the lowest) corresponds to three queue thresholds, onset, discard and abatement. If an STP routes an MSU, with priority n, to a link whose transmission queue has exceeded the n+1th onset threshold, a TFC is returned to the originating SP. This indicates the current congestion level (e.g. 1 to 3). If the queue has also exceeded the n+1th discard threshold, then the MSU (which has priority n) is discarded. By selectively discarding messages, the load on the congested link is reduced. Also the priority scheme restricts the least important MSUs first. For example, MSUs which set up new calls have the lowest priority, as the emphasis during congestion is to complete those call attempts already in progress [RUM93]. MTP3 informs User Parts of the current congestion level (as indicated by either the local transmit queue or arriving TFCs), so that the User Parts may throttle MSUs whose priority is lower than this.

The alternate National congestion control scheme, without MSU priorities, has just two congestion thresholds, onset and abatement. The congestion level (or state) is incremented for every T_x seconds that the onset threshold is exceeded, and decremented for every T_y seconds that the queue is below the abatement level (T_x and T_y are implementation dependent). The congestion state is indicated in TFC messages, as with the priority scheme. However in this instance, the congestion state indicates only the level by which the User Parts should decrease their transmission rate, rather than specifying which MSU types should be throttled.

Both National congestion control schemes have explicit procedures for SPs to determine congestion abatement. While congestion is active, MTP3 sends "Signalling-route-set-congestion-test" (SRSCT) messages, which are assigned a priority one less than the current congestion state. These messages are sent, in the first instance, if no TFC has arrived for T_{15} seconds (i.e. 2-3 seconds). If the congestion state is the same, or has increased, then the SRSCT message will trigger a TFC in the reverse direction which indicates this. If the congestion state has gone down, there is no response. In this case the SP waits T_{16} seconds (i.e. 1.4-2 seconds), decrements the congestion state, then sends another Signalling-route-set-congestion-test. This continues until the congestion state has returned to 0.

Having described the operation of the SS7 congestion controls, we now review previous analyses of their performance. [LEE89] models an earlier version of the national scheme, without priorities. The purpose of this study is to determine the steady state message throughput and delay (i.e. during sustained congestion) as a function of the congestion thresholds and the timers T_{15} and T_{16}.

[MAN94] examines performance, during sustained overload, of a congestion control scheme with priorities, by means of a simulation study. The aim is to determine the effect of various congestion thresholds on a) the discard probability of low priority messages and b) the frequency at which the congestion controls are invoked. While this frequency is said to determine the overall message throughput, actual throughput results are not given.

[ZEP94] presents a simulation study of the transient performance of both the International and the National (i.e. with priorities) congestion control schemes. The results show that when the number of SPs affected by congestion controls increases (e.g. from 10 to 1000), the TFC mechanism is unable to prevent traffic from overflowing at STPs.

congestion controls no longer work correctly. Incorrect operation in this context means that, despite the congestion controls, messages are discarded at the affected STP.

While the aim of the above studies is to determine signalling message throughput during congestion, [RUM93] [RUM94.1] [RUM94.2] and [SMI94] consider instead call completion rate. In particular, [RUM93] shows examples where the message throughput (for a given congestion scenario) is good, but however the call completion rate is very poor. The conclusion is that call completion rate is better than throughput for indicating performance during congestion. [RUM93] considers overload at level 3 of an STP (previous studies have considered level 2 congestion only), and models a priority scheme where messages are discarded progressively by the STP (messages relating to new call attempts have lowest priority, and are discarded first). With the scheme described in [RUM93], no control (i.e. via TFC messages) is exerted on offending SPs.

[RUM94.1] presents a simulation study of the same scenario, i.e. congestion at STP level 3, however in this case the national congestion control scheme is modelled. Here TFC messages are sent by the congested STP. Results show that (not surprisingly) call completion rates are very poor if a TFC is sent for each MSU arriving during congestion, due to the overhead imposed by the TFCs. The results also show that the generation of TFCs is oscillatory (i.e. they occur in periodic bursts). [SMI94] examines MTP level 2 congestion, and notices similar oscillatory behaviour. [SMI94]suggests reducing the value of timers T_{15} and T_{16} to eliminate these oscillations.

The above papers show that SS7 congestion control is an active research area, with further avenues for research identified in [RUM94.2]. While we have reviewed this area, to allow a complete overview of the SS7 MTP, we have not pursued congestion control in the B-ISDN MTP in this project. Our aim instead has been to identify the differences in the B-ISDN MTP arising from the use of ATM VCCs, and to pursue the engineering questions which arise from these differences. While the increased bandwidth of ATM VCCs (compared to SS7 links) will no doubt influence congestion control procedures, this effect could easily be incorporated into the above studies by changes in parameter values.

The SS7 protocols described so far in this chapter are designed to provide reliable connectionless message transfer over 64 kbps circuits. The advent of B-ISDN (and in particular ATM) provides a fundamentally different paradigm for signalling message transfer. Here Virtual Channel Connections, of variable bandwidth, can be provided at relatively low cost (compared to the current 64 kbps circuits). In addition, the different protocols used at the link level will result in a major changes to error monitoring and performance modelling procedures (as will be shown in subsequent chapters).The next section reviews B-ISDN, and outlines the ATM transfer technique and the protocols used to send signalling messages on an ATM network.

2.6 B-ISDN Overview

ISDN has been the focus of much effort in the ITU-TS standards bodies over the last decade. The ISDN paradigm is to have a single network, and a single access protocol, which supports a wide range of services, both voice and non-voice. ISDN uses 64 kbps circuits to provide subscribers with a set of standard access rates. These range from basic rate (two 64kbps user channels, one 16 kbps signalling channel) to primary rate (thirty 64 kbps user channels, one 64 kbps signalling channel) [STA92]. (The US primary rate access is twenty three 64 kbps user channels, and one 64 kbps signalling channel).

The advent of fibre optic transmission systems has made services such as distributed video and image transfer feasible. The bandwidth needed for these services, tens to hundreds of Mbps [LYL92], is much higher than the ISDN primary rate. In addition, the widening use of Local Area Networks, and the need to interconnect these, means that there is an immediate demand for higher bandwidths than offered by the ISDN basic and primary rates. This has led to the development of Broadband ISDN (B-ISDN), which is designed to provide these higher bandwidths while maintaining the ISDN principle, i.e. a single network and access protocol for all services. B-ISDN is defined by the ITU-TS as a "Service requiring transmission channels capable of supporting rates greater than the primary rate" [I.121].

2.6.1 ATM

While the bandwidth offered by fibre optic systems makes B-ISDN possible, a transfer mechanism is needed to allow different services to access (and share) this bandwidth. The target transfer mechanism specified by the ITU-TS is the Asynchronous Transfer Mode (ATM). As the title indicates, this is a fundamentally different technique from the Synchronous Transfer Mode (STM) used by ISDN. Here we describe the ATM technique and the protocols which will allow signalling traffic to be carried on a B-ISDN network.

The current digital network uses the Synchronous Transfer Mode (STM) to allocate bandwidth between users. STM transmits fixed length frames at regular intervals, each frame having a set of 64 kbps time slots. A call is assigned one slot per frame (or more depending on the bandwidth needed), with the same slot (or slots) being used for the duration of the call. This scheme is unsuitable for B-ISDN, for two reasons. Firstly, an STM multiplexor can only offer a fixed set of channel rates. This is inefficient for services whose requirements do not match these rates. Secondly, once bandwidth has been assigned, it remains so for the duration of the call. This results in idle capacity when services are bursty.

The Asynchronous Transfer Mode (ATM) is designed to fix these shortcomings, by providing a wide (and variable) range of bandwidths while carrying bursty services efficiently. ATM divides all traffic into fixed length, self routing cells. Due to the limited functionality of ATM switching (compared to X.25 for example), ATM cells can be switched at very high speed. Commercially available switches now offer throughputs of over 1 Gbps [BIA93].

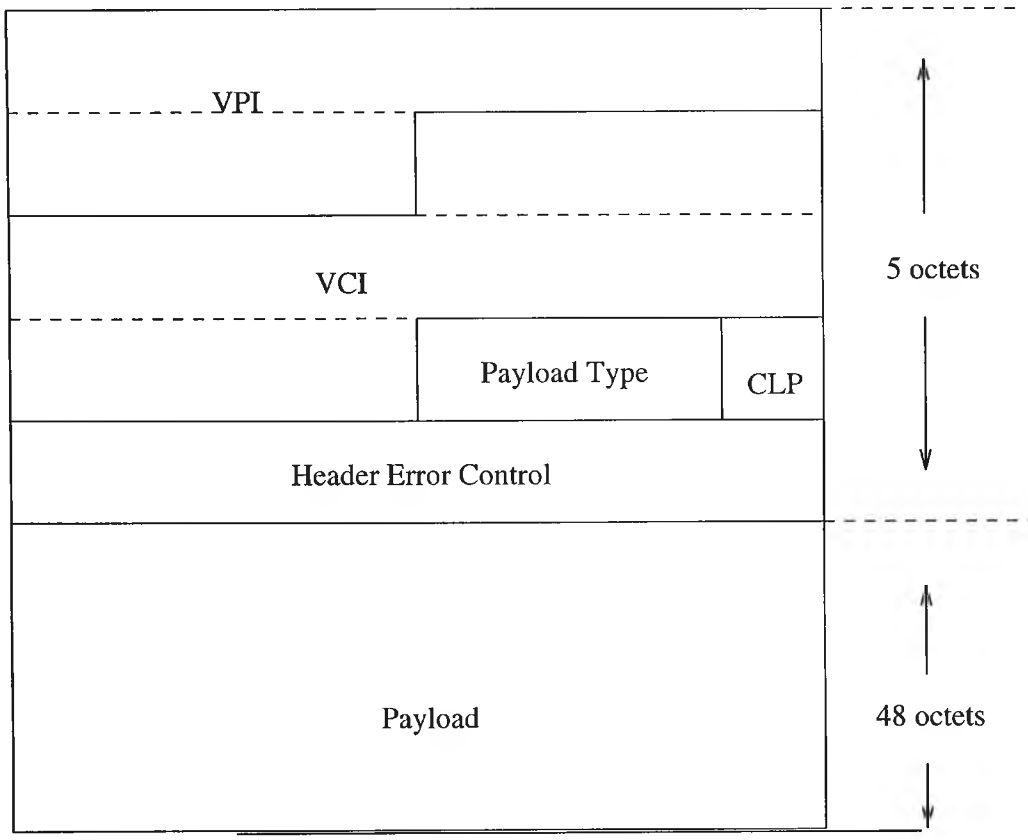


Figure 2.6 ATM Cell Structure

The basic ATM transmission unit, the cell, allows 48 octets for user data with a 5 octet header. This header, shown in Figure 2.6, has a one octet field which does error detection and single bit error correction. This protects the remaining 4 octets. The payload type field is used to identify the cell type (management or data), and provides a congestion indicator. This is the Cell Loss Priority (CLP) bit, which allows 2 levels of cell priority. Low priority cells (those with the bit set) may be discarded first when congestion occurs.

ATM is connection oriented. Logical connections (known as Virtual Channel Connections (VCCs)) are established between end users, similar to the Virtual Channels used in X.25 [STA89]. These ATM VCCs have a two part address field, as shown in Figure 2.6. This comprises a 16 bit Virtual Channel Identifier (VCI) and a 12 bit Virtual Path Identifier (VPI). At the User Network Interface (UNI), the VPI field is 8 bits (Figure 2.6 shows only the Network Network Interface cell structure). ATM addresses provide 4096 Virtual Paths (VPs), each with 65536 Virtual Channels (VCs). Switching is done at two levels, the Virtual Path and Virtual Channel level (some ATM switches, known as Virtual Path Cross-connects or Virtual Path Switches, switch Virtual Paths only [DEP92]).

The ability to route a large bundle of Virtual Channels by routing a single Virtual Path allows simpler call establishment and network management. Call establishment becomes easier, as bandwidth for a VCC need only be assigned once, at the entry to the VP. Without an underlying VP, a VCC must negotiate bandwidth at each switch, which increases connection setup complexity [BUR91]. VPs simplify network management in two ways. Firstly, failure recovery can be achieved by re-routing VPs, rather than re-routing each VC within a failed VP. Secondly, services can be separated on a Virtual Path basis [ANI89]. This would allow separate overlaid "Virtual Service Networks", which operate independently (and without mutual interference), due to policing of the bandwidth allocated to each VP [ATK93].

ATM has built in performance monitoring (or Operations and Maintenance (OAM)) capabilities. These OAM procedures, which operate at both the Virtual Path Connection (VPC) level and the VCC level, are described in ITU-TS recommendation I.610 [ITU93.4]. This ability to monitor individual VCCs using OAM cells raises the question: can an ATM signalling error monitor scheme be designed which uses OAM information, rather than counting errored Signals Units (the SUERM approach) or errored time intervals (the EIM approach)? This third error monitoring possibility (i.e. using OAM information) will be explored further in chapter 5. As a prelude to this, we shall describe how the OAM procedures work.

The ATM OAM procedures used dedicated cells for VCC performance monitoring and reporting. These have a special payload type field in the ATM cell header, which distinguishes them from user cells. VPC OAM cells are carried on dedicated OAM VCCs within a VPC.

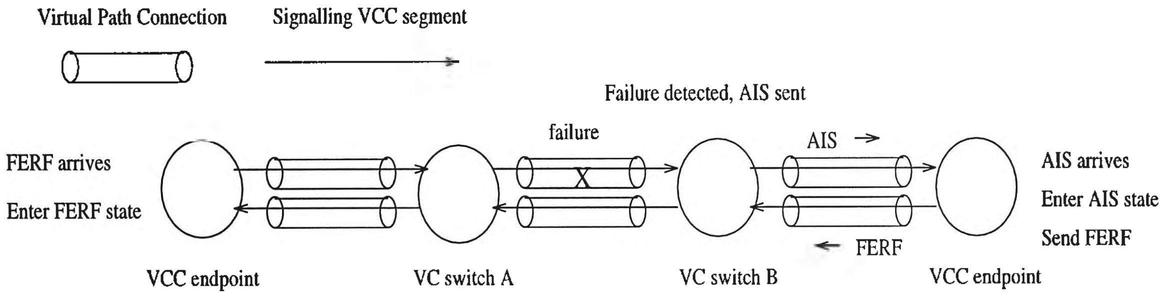


Figure 2.7 Failure Indication Using OAM

When a VCC failure is detected, an OAM Alarm Indication Signal (AIS) cell is sent by the transit VC switch which detects the fault. This is shown in Figure 2.7, where VC switch B sends the required AIS cell after detecting the fault (note that both directions of the VCC are shown). These AIS cells are sent continuously while a fault is pending, at a nominal rate of 1 cell/sec. When the AIS cell arrives at the VCC endpoint, an OAM Far End Receiver Failure (FERF) cell is sent in the reverse direction, as shown in Figure 2.7 (FERF cells are also sent continuously while faults are pending). This exchange of AIS and FERF cells is required for both ends of the VCC to learn of the failure. Clearly a failure in both directions of the VCC means that the FERF cells would not arrive. In this case the failure would be advertised by sending AIS cells in both directions.

To monitor the error performance of a VCC (or VPC), an OAM performance monitoring cell is inserted after each block of user cells. The minimum block size is 128 cells. This OAM performance monitoring cell indicates the number of cells in the block (to allow cell loss detection), as well as providing a parity check over the block. When the link becomes idle, OAM continuity cells are sent at regular intervals (Tc). If no continuity (or user cells) are received during an interval greater than 2Tc, then the VCC (or VPC) is presumed to have failed.

The ATM technique provides an efficient transfer mechanism, based on fixed length cells. ATM must interface with higher layers, which of course are not based on fixed length cells. Before considering this interface however, we will describe briefly the physical layer, which will carry ATM cells.

2.6.2 SDH

The Synchronous Digital Hierarchy (SDH), is the international standard for fibre optic communication[2]. Originally proposed by Bellcore, it was standardised by the ITU-TS in 1988 [BAL89]. SDH is compatible with current European and US transmission formats (such as E-1, DS-1), as well as being able to carry ATM cells. A major aim of the standard is to allow the interconnection of SDH equipment from different vendors [OMI93].

The SDH building block is the STM-1 frame, which operates at 155.52 Mbps. STM-1 frames can be concatenated to give higher rates, with STM-4 (622 Mbps) and STM-16 (2.4 Gbps) currently standardised. SDH differs substantially from the current transmission standard (Plesiochronous Digital Hierarchy (PDH)), as individual lower bit rate streams (known as Virtual Containers) can be extracted from STM frames, without disassembling the entire frame. Each Virtual Container has a pointer assigned to it, which indicates the location of the Virtual Container within the SDH frame. ATM cells are carried in “VC4” Virtual Containers, which occupy the entire payload of an STM-1 frame [SEX92].

SDH provides multi-level performance monitoring. This allows bit errors to be monitored over the entire STM frame, as well as within individual Virtual Containers. In addition, a standard method is specified for line level (i.e STM) Protection Switching. This allows SDH systems to recover automatically from line level failures (e.g. cable cuts, regenerator failures).

2. This standard, with some minor differences, is known as SONET in the USA.

The combination of SDH and ATM will provide cell based transport over fibre optic systems. This will carry Virtual Channel connections with differing (and variable) bandwidth requirements. To see how applications (such as signalling) will interface to an ATM network, we now consider the B-ISDN Protocol Reference Model.

2.6.3 B-ISDN Protocol Reference Model

The Physical Layer, which encompasses SDH, and the ATM Layer form the two bottom layers of the B-ISDN Protocol Reference Model [I.321], shown in Figure 2.8. The higher layers shown in the figure (which correspond roughly to OSI Layer 3 and up) are divided into two planes. The Control Plane refers to call and connection control (i.e. signalling), while the User Plane corresponds to user data transfer. Layer Management performs management functions for individual layers, an example being the establishment of signalling Virtual Channels (a procedure called metasignalling). Plane Management performs management functions relating to the system as a whole, and provides coordination between all planes.

The ATM Layer provides uniform cell based data transfer for all services. However these services will have widely differing data formats and requirements (e.g. error correction, timing). The purpose of the ATM Adaptation Layer (AAL) is to interface different services to the ATM Layer. Four separate AALs have been defined [I.363]. These are:

- AAL1: Constant bit rate services which require timing (e.g. circuit emulation)

- AAL2: Variable bit rate services which require timing

- AAL3/4: Variable bit rate services without timing (e.g. data services). The AAL3/4 allows individual data streams to be multiplexed within a single VCC, by providing a Multiplexing Identifier (MID) field in each cell.

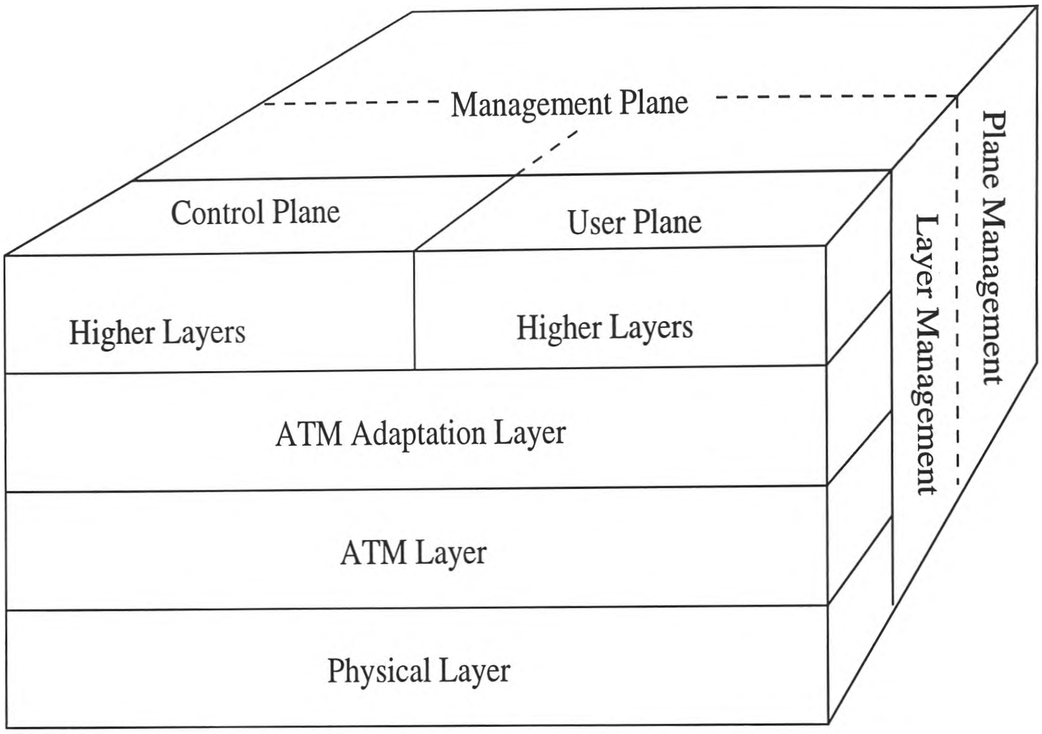


Figure 2.8 B-ISDN Protocol Reference Model

- AAL5: Variable bit rate services without timing, and without the AAL3/4 multiplexing capability.

As the AAL5 is to be used for signalling (as well as other B-ISDN services), we will outline it in more detail. Initially called the Simple and Efficient Adaptation Layer (SEAL) [WAN92], the AAL5 delineates Protocol Data Units (PDUs) and provides error detection with minimal overhead (compared to AAL 3/4). A single bit within payload type field of the ATM cell header is used to indicate the last cell of a PDU (so that, by implication, the next cell starts a new PDU). There is no PDU header, instead each PDU has a trailer which has a length indicator and a 32 bit CRC. The maximum AAL5 PDU length is 65536 octets [I.363].

The functions for each AAL fall into two groups, the common functions (e.g. PDU segmentation and reassembly) and the service specific function (e.g. error control). The common functions for the signalling AAL are done by AAL5. The service specific functions are outlined in the next section.

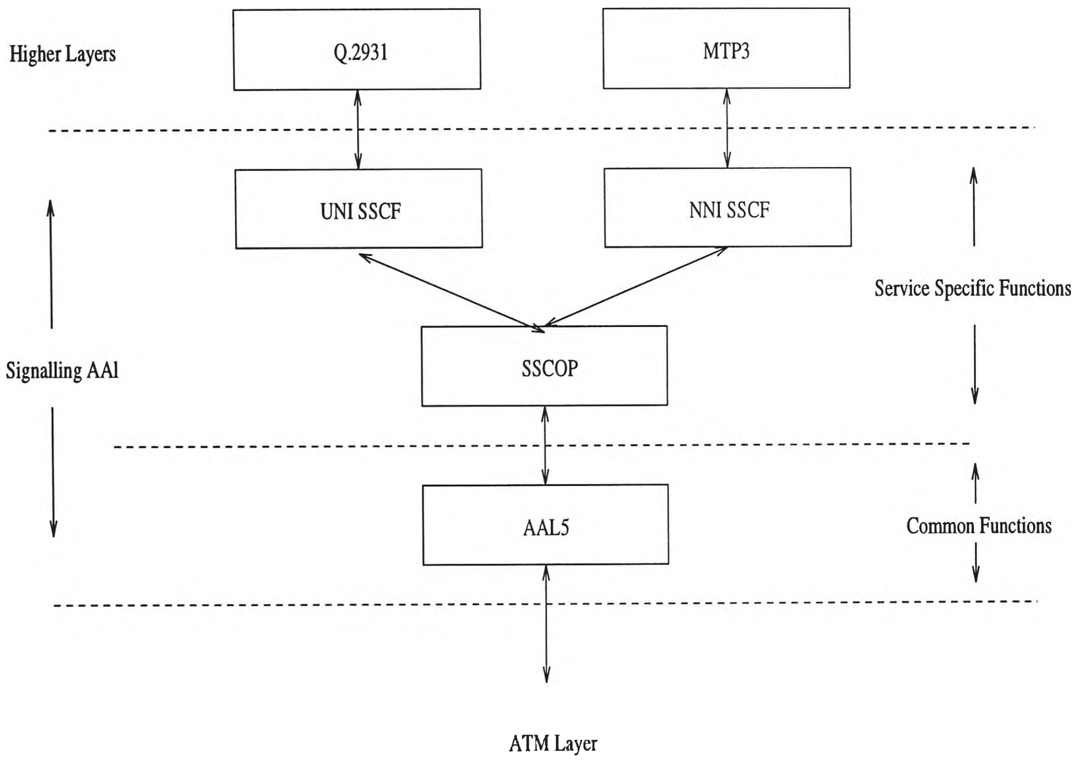


Figure 2.9 Signalling AAL

2.6.4 Signalling ATM Adaptation Layer

The Signalling AAL (SAAL) provides a roughly equivalent service to (and is intended to replace) MTP2. The lower half of the signalling AAL comprises AAL5, as mentioned. The remainder of the SAAL is described here.

The service specific part of the signalling AAL (known as the Service Specific Convergence Sublayer (SSCS)) is subdivided further, as shown in Figure 2.9. The Service Specific Connection Oriented Protocol (SSCOP) provides connection establishment and error control, similar to link level protocols such as HDLC. There are major differences however; the SSCOP uses Selective Repeat error correction [TAN88], with acknowledgements carried in periodic "Status" cells [Q.2110]. The operation and performance of the SSCOP is described in detail in chapter 4.

The Service Specific Coordination Function (SSCF) maps the SSCOP to the upper signalling layers. Two SSCFs are defined, one for the User Network Interface [Q.2130], the other for the Network Network Interface [ITU94.5].

The User Network Interface (UNI) SSCF serves the UNI signalling protocol, Q.2931 (otherwise known as Q.93B or DSS2). This SSCF provides minimal added functionality, as it merely transfers PDUs between Q.2931 and the SSCOP. The Network Network Interface (NNI) SSCF however is far more complex, as it interfaces the SSCOP to MTP3. Intended functions include congestion indication, PDU (i.e. MSU) retrieval during changeover, and link (i.e. VCC) proving. These functions are still under study (as indeed is the use of MTP3 itself). In chapter 6 we examine the NNI SSCF and the B-ISDN MTP3 in detail.

2.6.5 B-ISDN Call Control

Previous sub-sections have introduced the ATM layer and Signalling ATM Adaptation Layer. Here we describe briefly the protocols for B-ISDN call control.

There are two main bodies involved in B-ISDN standards activities. The first is the ITU-TS (International Telecommunication Union Telecommunication Standardization Sector, formerly the CCITT). The second is the ATM Forum, a consortium of equipment manufacturers. While the ATM Forum does not produce standards, it has accelerated substantially the pace of standards activity through the release of an ATM UNI specification [ATM93] and the development of a public NNI specification (known as the Broadband Intercarrier Interface (B-ICI)). These two specifications, which define the physical layer, ATM layer, AAL and the signalling protocols for call control, reuse ITU-TS standards where-ever possible. The signalling protocols for call control fall into two categories, UNI call control and NNI call control. We begin with the UNI call control protocols.

To keep up with the accelerated pace of standards development, the initial ATM UNI call control protocol, Q.2931, has been based closely on its ISDN counterpart (Q.931). Q.2931 provides the Layer 3 call control signalling capabilities for Capability Set 1 (CS-1) (sometimes referred to as Release 1)

[3] In particular Q.2931 provides the capability to establish Point to Point Class A,C and X ATM VCCs (Class A connections are Constant Bit Rate (CBR) with timing requirements, Class C connections are Variable Bit Rate with no timing requirements. For Class X connections, the AAL type, traffic type (i.e. VBR/CBR) and timing requirements are user defined, and are transparent to the network). The ATM Forum has adopted an earlier version of Q.2931 (known as Q.93B) for the User Network Interface 3.0 [ATM93]. However a notable difference is that, unlike Q.2931 (and CS-1), the ATM UNI 3.0 specification supports uni-directional Point to Multi-Point calls. Version 3.1 of the ATM Forum UNI is aligned with Q.2931 (instead of Q.93B).

Similar to the UNI standards, the NNI signalling protocols have reused much of the existing ISDN call control signalling protocols, known as ISUP [ITU92]. Accordingly, the resulting protocol is known as B-ISUP (i.e. Broadband ISUP) [ITU94.3]. Initially there was an attempt to align the UNI and NNI signalling protocols, so that B-ISUP and Q.2931 would have been identical [DIN93].While this may have simplified implementations, the process was abandoned, as it was unlikely to produce the required signalling protocol in time. However B-ISUP is largely similar to Q.2931, the major differences being some additional management functions in B-ISUP (e.g. VP blocking/unblocking, User Part unavailable indication etc.).

The introduction of new call types, other than the simple Point-to-Point calls, will lead to a separation between Call Control and Bearer Control. With the current call control procedures (i.e. ISUP), a bearer (i.e. a circuit) is setup in conjunction with the call control signalling. However the new protocols proposed for CS-2 signalling [ITU93] will provide separate message flows for

3. B-ISDN Signalling Capability Set 1 (defined by ITU SG 11) and Release 1 (defined by ITU SG 13) refer to an almost identical set of capabilities, i.e. the ability to establish Point to Point ATM VCCs with asymmetric Bandwidth Requirements [ITU94.2] B-ISDN Signalling Capability Set 2 (CS-2), which is the next proposed step in B-ISDN development, will allow Point to Multi-Point calls and renegotiation of Traffic and Quality of Service during the active phase of a call [ITU94.3]. CS-2 is a subset of the Release 2 capabilities

call and bearer establishment (with ISUP, a single message (the Initial Address Message (IAM), does both). While B-ISUP messages combine the call and bearer control functions. there has been an attempt to separate these functions in the protocol specification.

2.7 B-ISDN Signalling Transport

The preceding SS7 and B-ISDN overviews lead to a major issue, namely which architecture is most appropriate for a B-ISDN signalling network. This section describes three potential B-ISDN signalling architectures. We begin by identifying the main categories of signalling traffic to be carried by these architectures.

SS7 traffic falls into two main categories: circuit and non-circuit related. Initially all SS7 traffic was circuit related, so that it was not possible for Signalling endpoints to communicate without setting up a circuit at the same time. To allow non-circuit related signalling (for example queries to IN databases for 008 (freephone) number translation), the SCCP and the Transaction Capabilities Application Part (TCAP) were introduced in the 1988 standards [CLA88].

SS7 call control, done by ISUP and TUP, is combined with circuit (or bearer) setup. Separate call and bearer control however is planned for B-ISDN CS-2, as mentioned in the previous section. This allows more efficient use of network resources; for example the availability of a called party can be established before bandwidth is assigned for a call.

We therefore identify three distinct categories for B-ISDN signalling messages:

i) Bearer Control messages, between the ATM Virtual Channel switches along the route of a VCC.

ii) Messages not directly related to a call or bearer control, e.g. IN database queries.

iii) Call control messages, between signalling end points.

Due to this third signalling message category, all B-ISDN signalling endpoints will exchange messages directly. The network must therefore provide (and be dimensioned for) this capability. While the SCCP allows SS7 messages to be exchanged between signalling endpoints, most SS7 messages travel between adjacent SPs, for circuit setup.

With this requirement for end to end message transfer in mind, we now examine alternatives for B-ISDN signalling transport.

2.7.1 MTP

One method for transporting B-ISDN signalling messages is to use the current SS7 network. This is one of the alternatives considered in [SKO89]. There are two disadvantages with this approach. Firstly, while MTP2 restricts messages to 272 octets, CS-2 messages are expected to be much longer [ITU93]. Secondly MTP2 places an upper limit on the signalling capacity between two SPs, due to the limit on the number of links in a linkset. By contrast, an ATM signalling VCC has no a priori restriction on capacity, nor message size (however AAL5 restricts messages to 64k octets). Due to the ready availability of the ATM network for carrying signalling traffic, and the preference for this method expressed in the B-ISDN signalling standards, using the SS7 network (i.e. MTP) for B-ISDN signalling will not be considered further in this thesis.

2.7.2 MTP3 and the Signalling AAL

This method replaces MTP1 and MTP2 with the ATM network and the Signalling AAL. However MTP3 remains unchanged. The current standards allow this approach, as the proposed Signalling AAL supports the current

MTP3 primitives [ITU94.5]. By retaining the MTP3 routing capability, this second approach allows the STP based SS7 network architecture to be used for B-ISDN signalling.

Carrying the SS7 architecture over to B-ISDN has advantages, as SS7 design techniques and problems are well understood [CAM92,KLE88]. Furthermore, using high speed ATM VCCs instead of 64 kbps signalling links reduces the likelihood of congestion, which is due mainly to insufficient transmission capacity [MAN93.1]. However the SS7 architecture was designed for 64 kbps signalling links, not ATM VCCs. It is therefore necessary to re-examine the assumptions which have led to the current SS7 architecture, to see if they still apply for ATM.

2.7.3 Fully Meshed

A major reason for deploying STPs in SS7 networks has been to use fewer signalling links, thereby reducing transmission and termination costs [HIL88,STA87]. However the low cost of ATM signalling VCCs, compared to SS7 signalling links [GID93], reduces these savings. Hence a fully meshed B-ISDN signalling network, without STPs, cannot be immediately discounted on the basis of signalling link cost.

By eliminating STPs, a fully meshed B-ISDN signalling network removes a potential congestion and failure point and eliminates STP costs. Against these benefits must be weighed the additional bandwidth, control and memory costs of a fully meshed (compared to an STP) network. In addition the delay performance and failure recovery capability of these two approaches must be compared.

2.8 B-ISDN MTP

B-ISDN signalling will require a service similar to the SS7 MTP, to transport signalling messages. However to implement an MTP in an ATM environment, some major engineering issues, arising from the differences between ATM VCCs and SS7 links, will need to be resolved. This section outlines these issues, drawing on the background introduced in the earlier sections. The solutions we propose occupy the remainder of the thesis.

2.8.1 Differences between SS7 and ATM Signalling

The previous sections have reviewed SS7 signalling and B-ISDN. This has highlighted some clear differences between SS7 and ATM signalling, which will impact the development of a B-ISDN MTP. We summarise these differences here.

1) SS7 signalling uses go-back-n error correction, whereas ATM signalling will use Selective Repeat.

2) SS7 links maintain a constant bit rate, due to the Fill In Signal Units (FISUs) which are sent during idle periods. It is unlikely however that FISUs will be used on ATM signalling links, so that the bit rate on these links will vary with the traffic load.

3) The signalling peak rate for SS7 is fixed, usually at 64 kbps. To obtain additional signalling capacity, multiple links are used in parallel. In contrast, a wide range of peak rates is possible for ATM signalling links, due to the bandwidth flexibility of ATM.

4) SS7 signalling architectures are STP based. A major reason for this is to reduce the number of signalling links. While ATM signalling architectures have yet to be determined, the low cost of ATM signalling VCCs (compared to SS7 links) may make possible alternate approaches (e.g. fully meshed).

5) SS7 signalling protocols are self contained. In contrast, ATM signalling will use a number of general purpose ATM components, e.g. the ATM layer, AAL5 and the SSCOP. These have some built in error monitoring capabilities, as well as having different procedures for error detection than MTP2

6) Due to the more complex (than SS7) call types planned for CS-1 and CS-2, MSU lengths in ATM signalling systems will be longer than their SS7 counterparts.

2.8.2 B-ISDN MTP Design Issues arising from these differences

The differences between SS7 and ATM signalling outlined above raise some major design issues for a B-ISDN MTP. These issues are outlined here, then addressed in the remainder of the thesis.

As the delay performance of the signalling network directly affects call setup times, the control of signalling link delay is a key requirement. There is a clear need to model this delay, and the amount that it increases during error periods. While SS7 delay performance has been modelled extensively, the results do not apply to ATM signalling links, due to the use of Selective Repeat in ATM. Accordingly existing Selective Repeat (as well as SS7) performance studies should be reviewed, leading to the development of a new delay performance model for ATM signalling. This model should incorporate the operation of Selective Repeat, as well as the other features of the SSCOP protocol (e.g. the error reporting procedure).

In addition to revising techniques for delay performance modelling, a new approach to error monitoring is needed. This is due to the differences between SS7 links and ATM VCCs outlined above, in particular the variable bit rate on ATM signalling links, the availability of performance monitoring capabilities in the SSCOP and the OAM layer, and the different error detection procedures of the Signalling AAL (compared to MTP2). In addition, there is an opportunity to rectify any deficiencies with current SS7 error monitoring schemes. The previous sections have identified three possible error monitoring schemes, which are:

i) The SUERM (i.e. the Leaky Bucket).

I) The Errored Interval Monitor (EIM).

iii) A scheme based on OAM measurements from Signalling VCCs.

Each of these schemes needs to be evaluated in an ATM environment, to find the most effective error monitor for ATM signalling. This evaluation should account for the factors outlined above, as well as previous studies of SS7 error monitoring performance.

Due to the low cost of ATM signalling VCCs, compared to SS7 links, a fully meshed ATM signalling architecture may be feasible. However an investigation is needed of the increased bandwidth requirement (compared to an STP based network), as well as the respective call handling capacities of the two signalling architectures. A comparative study of these two potential ATM signalling architectures (i.e. fully meshed and STP based), which considers these two aspects, is therefore needed.

Assuming that fully meshed ATM signalling networks may be deployed, it is necessary to examine their MTP protocol requirements (in addition to the need for an error monitor). From the MTP3 description given in this chapter, it is evident that a major task of MTP3 is to ensure the integrity of STP based networks. However if STPs are not deployed, a simplified MTP3 can be used. Accordingly a close examination of the protocol functions needed for this simplified B-ISDN MTP is needed. This should consider both MTP3 (focusing on the needs for fully meshed networks) and MTP2. In particular, to complement the work on ATM error monitoring, the MTP2 component should consider the ATM equivalent of the SS7 Link Alignment procedure.

3. Signalling Performance Modelling and Error Monitoring Review

3.1 Introduction

The previous chapter identified error monitoring as a major requirement for a B-ISDN MTP. An key input to the error monitor design will be the maximum error rate sustainable for ATM signalling links (i.e. the maximum error rate which still allows delay constraints to be met). To determine this maximum sustainable error rate, the delay performance of ATM signalling links under errored conditions needs to be studied.

This chapter reviews previous work on SS7 delay performance modelling and error monitoring. The aim is to determine the extent to which SS7 results may be applied to the B-ISDN MTP, and to identify where new work is needed. In addition, a critical review of previous Selective Repeat analyses is presented, as Selective Repeat will be used on ATM Signalling Links. The deficiencies identified with these Selective Repeat analyses are addressed in chapter 4, where a new model for the delay performance analysis of ATM signalling links is described.

Section 3.2 presents the ITU-TS SS7 delay formulae, together with current SS7 performance modelling approaches. Extensions to the ITU-TS formulae, which give SS7 performance at very high error rates, are also outlined. Section 3.3 reviews previous Selective Repeat analyses, to determine their applicability to ATM signalling.

Section 3.4 reviews previous work on SS7 error monitoring, both for 64 kbps links (the SUERM) and for 1.536 Mbps links (the EIM). From this emerges the requirements for SS7 error monitors, as well as procedures for determining error monitor performance. This provides the background for Chapter 5, which examines how these error monitoring requirements may be met in an ATM environment.

3.2 SS7 Delay Performance Modelling

This section examines SS7 delay performance modelling approaches, beginning with a review of the ITU-TS SS7 delay formulae.

3.2.1 Overview

The approximate delay formulae for SS7 links, given in ITU-TS recommendation Q.706, model the performance of the go-back-n protocol as implemented by SS7. We begin by reviewing go-back-n operation in SS7 (i.e. at MTP2), then present the ITU-TS formulae.

Go-back-n requires the receiver to discard all packets that arrive after an error has been detected, until a successful retransmission of the errored packet arrives [STA89]. Upon learning of an error, the transmitter backs up to the errored packet, and begins retransmissions from that point. Due to its simplicity, the go-back-n protocol is widely used (e.g. in HDLC, LAPB and LAPD).

The SS7 implementation of the go-back-n protocol in MTP2 is unique in two regards. Firstly, as described in chapter 2, Fill In Signal Units (FISUs) are sent continuously when there is no other signalling traffic (this effectively maintains a link utilisation of 100%). However these FISUs are not retransmitted if errored. Secondly, SS7 uses two bits in each Signal Unit Header, known as the Forward Indicator Bit (FIB) and the Backward Indicator Bit (BIB), to indicate when retransmissions are needed [Q.703]. HDLC, in contrast, sends Reject (REJ) frames when skips in sequence numbers are detected, to show that retransmissions are due. The SS7 scheme works as follows. Under normal circumstances, the value of the FIB in one direction equals the BIB in the reverse direction. When the receiver detects a sequence number skip (and the FIB equals the BIB of the reverse link), the receiver inverts the BIB on all Signal Units sent in the reverse direction. The transmitter, upon detecting the reversal of the BIB, commences go-back-n retransmission. In addition, the transmitter reverses the FIB in all subsequent Signal Units, so that it once again equals the value of the received BIB. If a retransmitted Signal Unit is lost, the receiver will detect this by noting a) a skip in sequence numbers and b) the same value of the FIB and the BIB. The receiver will invert the BIB once more, which starts the retransmissions again.

3.2.2 ITU-TS Delay Formulae

Approximate delay formulae for SS7 links, for both errored and non-errored operation, are given in ITU-TS recommendation Q.706. We begin by listing the assumptions on which these formulae are based. These are:

1) MSU arrivals are poisson.

2) MSU errors are independent, with the same error probability for each MSU (regardless of MSU length).

3) MSU service time (i.e. transmission time) has a general distribution.

4) MSUs take non-preemptive priority over Fill In Signal Units (FISUs).

5) FISUs arrive according to a Poisson distribution (the actual arrival distribution for FISUs is not Poisson, as they are sent only when there is no other traffic).

6) There is a constant delay for reporting errors (i.e. between message transmission and the arrival of a negative acknowledgement).

The previous sub-section showed the last assumption to be reasonable, based on the SS7 implementation of go-back-n. Using these assumptions, MSU delay is obtained by modelling the level 2 (i.e. MTP2) transmitter as a non-preemptive priority M/G/1 queue, where FISUs have the lowest priority [CHE82]. Before presenting the ITU-TS SS7 delay formulae, we introduce the following terminology:

p_m MSU error probability (constant, regardless of MSU size)

λ MSU arrival rate (MSUs/sec).

S_m S_{m2} First two moments of MSU service time.

S_f FISU service time (a constant)

Q_m Mean MSU queueing time, i.e between joining the queue and entering the server for the first time

T_L Round trip time, i.e. twice the propagation delay plus processing time at the receiver.

ρ_m Utilisation of transmitter queue by MSUs (= λS_m)

$t_l = T_L/S_m$

$E_1 = 1 + p_m t_l$

$E_2 = 1 + p_m t_l + 2$

There are two versions of the ITU-TS delay formulae, for error free and errored operation. These give the queueing delay (i.e. the time until an MSU enters the server for the first time) normalised by the message transmission time. We have re-arranged these to show the absolute (i.e. un-normalised) queueing delay (as it is the absolute delay which is of interest from the perspective of error monitoring). The error free version is

$$Q_m = S_f/2 + (S_m/S_{m2})\,(\rho_m/(2\,(1-\rho_m))) \qquad \text{(Eqn 3.1)}$$

To obtain the errored delay, the MSU service time is expanded to include the additional delay due to retransmissions. This modified service time, known as the virtual service time S_v [BUX80], is given as

$$\begin{aligned} S_v &= S_m + (p_m T_L + p_m^2 2T_L + p^3{}_m 3T_L + \ldots)\,(1-p_m) \qquad \text{(Eqn 3.2)} \\ &= S_m + (p_m T_L)\,/\,(1-p_m) \\ &\approx S_m + p_m T_L \end{aligned}$$

The approximation in the last line is motivated by the small value usually taken by p_m (e.g. < 0.01). Using this expression for the virtual service time, the delay during errored operation becomes (after some manipulation):

$$Q_m = S_f/2 + S_m(\,(\rho_m E_2)\,/\,(2\,(1-\rho_m E_1)\,) + E_1 - 1) \qquad \text{(Eqn 3.3)}$$

Details of the calculation are in [CHE82]. In addition to the approximation mentioned above, the errored delay is also under-estimated due to the omission of the additional MSU service time which occurs with each retransmission. (Note: this omission is not mentioned in [CHE82]).

3.2.3 Link Dimensioning and Maximum Sustainable Error Rate

Here we review SS7 performance modelling studies, to see how performance modelling results are used to dimension SS7 links and determine maximum sustainable error rates.

[SKO87] and [SKO88] use the ITU-TS formulae to determine maximum utilisations (ρ_{max}) for SS7 links while satisfying constraints for mean delay and delay sensitivity (with respect to link utilisation). As an SS7 link is dimensioned to carry the load of a failed link as well (in the event of a changeover), ρ_{max} is chosen so that these delay constraints are satisfied at a link load of $2\rho_{max}$. Results are presented which show that ρ_{max} decreases as message length, or message length variability, increases. This is examined further in

[GHA91], which shows that if 5% of the MSUs are of maximum length (i.e. 272 octets), then the link utilisation must be reduced to a low level (0.05) to satisfy delay constraints. The same effect (i.e. a reduction of the link carrying capacity) is noted for correlated messages, which may arise if a large data segment is segmented into individual MSUs, then sent (for example in a User to User Information (UUI) exchange). To reduce the impact of long/correlated messages on SS7 delay performance, [GHA91] suggests that external controls be applied to restrict the proportion of long MSUs admitted. [KOS94] proposes a credit manager algorithm to reduce the impact of correlated message arrivals on SS7 delay.

The procedures in [SKO87] and [SKO88] are applied to both errored and error free operation (for the error free case, the delay constraints are more stringent). These delay constraints are formalised in ITU-TS recommendation E.733 [ITU92.2], which gives maximum delay specification for (essentially) error free operation (i.e. BER of 1e-06) and errored operation. E.733 extends the delay constraints used in [SKO87] to include the 99th delay percentile and the 99th delay percentile sensitivity (with respect to utilisation). The delay figures for errored operation apply to worst case operating conditions, i.e. an error rate of 1/256 (as the SUERM will initiate a link changeover when the error rate exceeded 1/256, SS7 links will not operate at error rates higher than this). ρ_{max} must be chosen so that under these worst case error conditions, and at a load of $2\rho_{max}$, the delay figures are not exceeded. These figures are: mean delay 60 msec, delay sensitivity (wrt utilisation) 300 msec/erlang, 99th delay percentile 300msec and 99th percentile sensitivity (wrt utilisation) 1500 msec/erlang. E.733 suggests that sensitivity results be obtained from graphs of delay/delay percentiles vs utilisation.

The studies mentioned so far in this section are in the context of current SS7 links, where the maximum sustainable error rate is set by the SUERM parameters (as described in chapter two). However [HOU94] instead determines the maximum SS7 error rate for given delay constraints and a given utilisation (in contrast to the previous papers, where the maximum **utilisation** is determined). The purpose of the work of [HOU94] is to re-examine the SUERM, and hence the maximum error rate is the object of study (rather than being set by the standardised SUERM parameters).

To incorporate delay percentile constraints, [HOU94] models go-back-n delay percentiles (this is also done in [CHE86], by numerically inverting delay transforms). In [HOU94] the aim is to keep the 99.99%th (instead of the 99%th) delay percentile below 300 msec. [HOU94] establishes the maximum bit error rate which an SS7 link (with a utilisation of 0.4) may sustain while still satisfying this delay constraint. This bit error rate is termed the "maximum sustainable error rate" (note: delay percentile sensitivity is not considered). There are three key observations made in [HOU94], which are

1) The maximum BER for 56 kbps links is affected by the message size distribution, with the maximum sustainable BER going down as the message size increases. This effect is particularly apparent for short links

2) As the link speed increases to 1.536 Mbps, the maximum sustainable BER is largely insensitive to changes in the MSU size distribution. This is because the MSU transmission time has become small compared to the round trip interval (i.e. the delay before an errored Signal Unit is retransmitted).

3) For both link speeds, the maximum sustainable BER is a function of the link length (i.e. the propagation delay).

Maximum sustainable error rates, i.e. the maximum error rate which satisfies delay constraints for a given utilisation, will be considered from an ATM perspective in chapter 5. In particular we shall examine how the maximum sustainable error rate is affected by changes in link speed (i.e. peak rate) and message size distribution.

3.2.4 Extensions to ITU-TS Delay Formulae

There are a number of assumptions which underly the ITU-TS delay formulae. In this subsection extensions to these formulae are described, which remove some of these assumptions.

The ITU-TS formulae assume that all retransmissions are successful (i.e. the probability that an MSU requires $n > 1$ retransmissions, which is $p_m^n (1 - p_m)$, is taken to be 0). While this is reasonable for small values of p_m (i.e. < 0.01), this assumption needs to be re-examined when modelling go-back-n performance at very high error rates. In addition, the service time of retransmitted MSUs is not modelled in the ITU-TS formulae. The necessary corrections are given in [SKO87], where the virtual service time for an MSU requiring n retransmission attempts is given as $S = S + n (S + T_L)$, where S is a random variable representing the MSU service time. Assuming the probability of n retransmission attempts to be $p_m^n (1 - p_m)$, the mean virtual service time is given as

$$S_v = S_m / (1 - p_m) + T_L (p_m / (1 - p_m)) \qquad \text{(Eqn 3.4)}$$

The second moment of the virtual service time is given in [BRA90]. This is

$$S_{v2} = \frac{(S_{m2} (1 - p_m) + S_{m2}^2 (2p_m) + S_m T_L (4p_m) + T_L^2 p_m (1 + p_m))}{(1 - p_m)^2} \qquad \text{(Eqn 3.5)}$$

The mean queueing delay (with the ITU-TS simplifying assumption removed), is obtained by replacing S_m and S_{m2} in Equation 3.1 with S_v and S_{v2} respectively, to give

$$Q_m = S_f / 2 + (S_v / S_{v2}) (\rho_e / (2 (1 - \rho_e))) \qquad \text{(Eqn 3.6)}$$

where ρ_e is the effective utilisation, including retransmissions. This is λS_v The same method is used to develop the SS7 delay formulae given in [WAN91].

An additional assumption, which has not been questioned previously, is that all MSUs have the same error probability, regardless of their length. It is more likely however that the error probability will be a function of the MSU length. Indeed, the increasing use of UUI signalling and database queries for Intelligent Network (IN) calls, as well as the potential use of the signalling network for transferring OAM files [ZEP94] makes it likely that there will be a proportion of long MSUs in future signalling traffic. The go-back-n model can be extended to account for this effect.

To model go-back-n delay, while accounting for the different error probabilities for different packet lengths, we introduce the following terminology

b the bit error rate on the link (independent bit errors are assumed)

m_i the probability that an MSU has a length l_i bits $(1 \le i \le h)$, i.e. there are h different lengths.

S_i the service time of an MSU of length l_i

p_i the error probability of an MSU of length l_i., where $p_i = 1 - (1-b)^{l_i}$

The task is, as previously, to find the first two moments of the virtual service time (i.e. S_v and S_{v2}). By extending Equation 3.4 and Equation 3.5 we get

$$S_v = \sum_{i=1}^{h} m_i \left(\frac{S_i}{1-p_i} + \frac{T_L p_i}{1-p_i} \right) \qquad \text{(Eqn 3.7)}$$

and

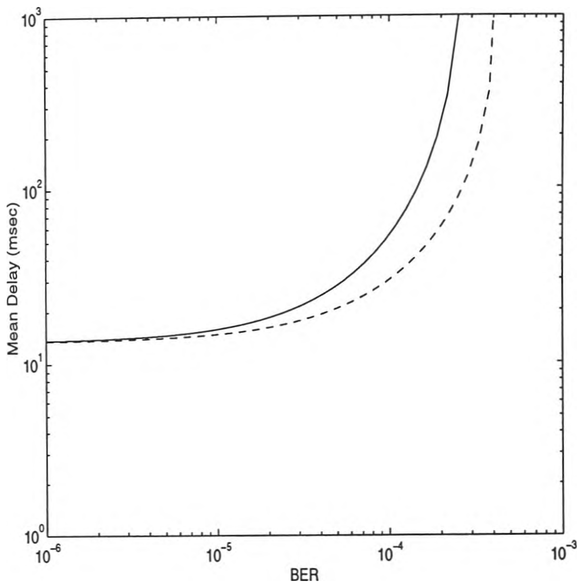


Figure 3.1 Mean delay vs BER for 64 kbps SS7 link with 90% 30 octet MSUs and 10% 272 octet MSUs. Dotted line result assumes constant error probability.

$$S_{v2} = \sum_{i=1}^{h} m_i \left(\frac{\frac{S_i^2}{1-p_i} + 2p_i S_i^2 + 4p_i T_L S_i + T_L p_i (1+p_i)}{(1-p_i)^2} \right) \qquad \text{(Eqn 3.8)}$$

The mean delay is obtained by substituting Equation 3.7 and Equation 3.8 into Equation 3.6.

Figure 3.1 shows the mean total MSU delay (i.e. $Q_m + S_v$) vs BER when the different error probabilities are modelled (the solid line), compared to the constant error probability model (the dotted line result). For this example, a link utilisation of 0.4 is used, with a round trip time (i.e. T_L) of 50 msec. As shown, the assumption of a constant MSU error probability (i.e. the mean error rate) causes the mean delay to be underestimated.

3.3 Selective Repeat

We review here Selective Repeat delay analyses, to examine their suitability for ATM signalling links.

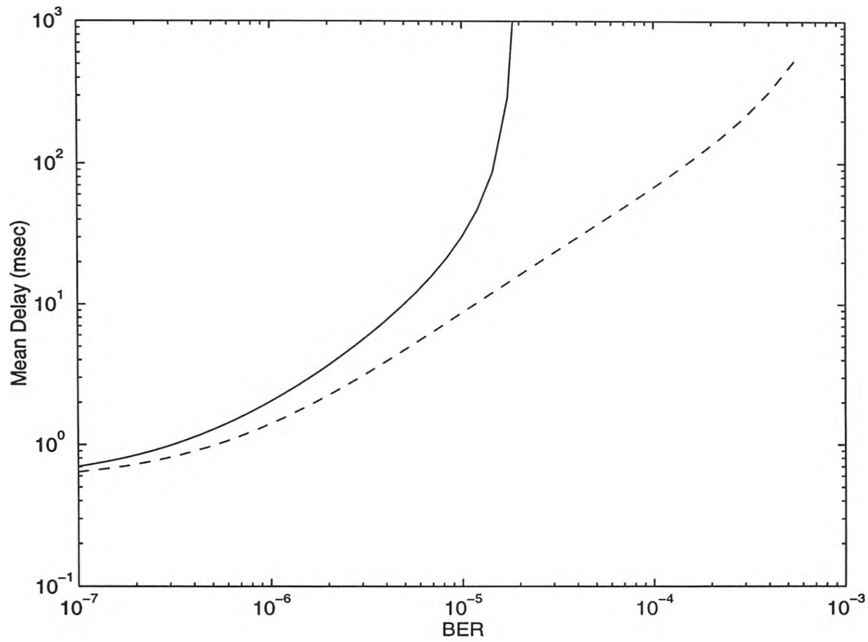


Figure 3.2 Comparison of Mean Delay vs BER for Selective Repeat (dotted line) and go-back-n (solid line)

We begin by considering the motivation for using Selective Repeat for ATM signalling. The new Signalling AAL will, in effect, be a replacement for MTP2. This has given an opportunity to choose the most effective ARQ procedure, without being constrained by existing implementations. While go-back-n was considered, the final choice was Selective Repeat. A major reason for this is shown by the delay comparison in Figure 3.2. Here we consider a high speed (i.e. 1.536 Mbps) link with the same traffic mix, link utilisation and round trip delay as Figure 3.1. The Selective Repeat delay (shown by the dotted line), obtained using the modelling procedures introduced in the next chapter, shows a marked improvement over the go-back-n delay (the solid line). This allows ATM signalling links (using Selective Repeat) to operate at much higher error rates than their SS7 counterparts.

Previous analyses of Selective Repeat delay performance fall into two groups, those concerning transmitter delay and those concerning receiver delay. Transmitter delay is measured from the time a packet first joins the transmit queue until a successful transmission is completed. However the Selective Repeat ARQ will also delay non-errored packets at the receiver if a retransmission of an earlier (errored) packet is still pending. This is so that the receiver may deliver packets in the correct order. Hence a study of Selective Repeat delay performance must consider both transmitter and receiver delay.

Transmitter delay for Selective Repeat is modelled in [KON80], [ANA86], [LEE91], [LU93], [YOS93]. These analyses all assume a constant round trip interval (even when multiple retransmissions are needed), and a constant packet size.The round trip interval has a length of S slots, where a slot is the packet transmission time. These assumptions (constant round trip time and packet size), which lead to this slotted approach, appear to have arisen while analysing the performance of satellite links [KON80] [YU81]. Using this model, [KON80] defines the system state at time t as

$$S(t) = (Q(t);r_1(t),\ldots,r_s(t)) \qquad \text{(Eqn 3.9)}$$

where $Q(t)$ is the transmitter queue length at time $t+1$ slots, and $r_i(t)$ equals 1 if a transmission was attempted at $t-i+1$ and 0 otherwise. [KON80] uses a generating function approach to obtain the transmitter queue length distribution and the virtual waiting time (i.e. the waiting time for an arbitrary packet), but notes that the approach becomes intractable for large round trip intervals (his results are for S = 4).

[ANA86] obtains Selective Repeat transmitter delay using two methods, both based on the model of [KON80]. The first is a Markov Chain approach, which becomes intractable for large S, due to the exponential increase in the state space as the round trip delay grows. However a simplified solution for transmitter queue length distribution (and hence transmitter delay) is provided for the case of geometrically distributed arrivals. The second method of [ANA86] assumes that the arrival of negative acknowledgements is independent of the past history of the system, thus ignoring the correlation between errored packets and their associated retransmissions. Results from this approximate method agree closely with simulation results, except when the error probability is very high and the round trip time short (the method underestimates the mean delay by about 10% at a packet error probability of 0.3 and a round trip delay of 5 slots). However the advantage of the approximate approach is that it remains tractable for long round trip intervals (previous methods, e.g. [KON80], do not), and provides a closed form solution when the arrival distribution is geometric.

To overcome the intractability of the first method of [ANA86], and the approximations made for the second method, [LEE91] introduces a new procedure for Selective Repeat delay calculation. While using the slotted approach of the previous methods, [LEE91] reduces the state space by condensing the information regarding previous transmissions (i.e $r_1(t),\ldots,r_s(t)$). Matrix geometric techniques are then used to obtain a queue length distribution. Results from [LEE91] overcome the inaccuracies of the approximate method of [ANA86], however these inaccuracies can be ignored unless the packet error probability is very high (e.g. 0.3).

[LU93] introduces a technique based on signal flow graphs [LU89], which models the effect of non-independent errors (all other methods assume independent errors). However as this approach only provides the delay once a packet has entered the server, and not the queueing delay, the results are of limited interest. [YOS93] analyses a scheme which divides messages into packets and transmits these packets using a Selective Repeat method. This effectively amounts to a stop and wait ARQ at the message level (each message waits until the previous message has been successfully transmitted), hence the results do not apply to signalling.

Selective Repeat delay in the receiver buffer (also called resequencing delay) is analysed in [ROS89]. This also uses the slotted approach employed in the transmitter delay analyses. Continuous traffic is assumed. [CHA94] uses a slotted approach to determine transmitter and receiver delay for an Adaptive Selective Repeat scheme, where unacknowledged packets are retransmitted during idle slots (even if no errors have been reported). This is similar to the

PCR scheme used in SS7. [PEY94] describes a scheme where the receiver sends status packets at regular intervals whenever a retransmission is pending. While [PEY94] claims to model receiver delay, only the time between an initial packet transmission and the arrival of an error free version of the packet at the receiver is calculated.

By assuming constant round trip intervals, these analyses fail to capture any variability in round trip delay introduced by the protocol operation ([PEY94] and [CHA94] do consider variations in round trip delays, but their results are inapplicable, for the reasons noted above). The effect of variations in round trip delay on Selective Repeat performance will be considered in chapter 4. In addition, the assumption of constant packet sizes leads to further inaccuracies. Firstly increases in queueing delay at the transmitter, due to variance in packet size, are neglected. Secondly, any possible effects on delay, due to changes in the packet error probability for different packet sizes, are not modelled (the assumption of constant packet size means a constant packet error probability).

This section has identified deficiencies in current approaches to Selective Repeat performance modelling. These deficiencies will be addressed in chapter 4.

3.4 Error Monitoring

Due to the critical nature of signalling traffic, and in particular the need to limit transmission delay for signalling messages, the error performance of SS7 links (and hence, indirectly, their delay performance) is monitored continuously. The purpose of this error monitoring is to detect when a link is no longer performing adequately (according to a specified error criteria), so that a link changeover can be done. The design of error monitors must address two criteria: firstly that a link changeover is done before an excessive queue of MSUs builds up (such a queue is called a changeover transient). Secondly, the error monitor must ensure that spurious changeovers do not occur (a spurious changeover is where the error monitor takes out an otherwise good link, due to an isolated error burst). This section reviews previous studies of SS7 error monitors with a view to:

a) Determining the requirements for SS7 error monitors (bearing in mind the criteria outlined above).

b) Identifying SS7 error monitoring schemes, to assess their suitability for ATM.

c) Identifying methods to determine error monitor performance (e.g. changeover time vs error rate).

Previous studies of SS7 error monitors consider three areas: changeover transients, time to changeover (as a function of error rate), and error monitor design for high speed (i.e. 1.536 Mbps) links. We examine these in turn.

3.4.1 Changeover Transients

Early studies of SS7 error monitors focused on the changeover transient. This transient must be controlled for two reasons: to limit the delay increase during changeover and to avoid triggering congestion control thresholds.

[AKI85] determines the mean and variance of the queue on the alternate link during a changeover, as a function of the rate at which MSUs are transferred from the failed link. By assuming this queue length to be normally distributed during changeover, congestion thresholds may be determined which, with a given probability, will not be triggered by changeover transients. The aim is to ensure that the link changeover procedure does not interfere with the MTP3 congestion controls.

[MAN93.2] considers the same problem (i.e. setting congestion thresholds which are not triggered by changeover transients), but assumes that messages are transferred instantaneously (i.e. in a single batch) from the failed link to the new one. The probability of reaching a given congestion threshold, once this batch arrives, is obtained by modelling the queue on the alternate link as a random walk. However to do this, it is necessary to assume exponentially distributed MSU lengths (the results of [AKI85] apply to generally distributed message lengths).

Both [AKI85] and [MAN93.2] assume that the amount of data queued on the failed link at changeover (i.e. the changeover transient) is known. [SKO87] obtains this transient by means of a simple model for the changeover time. This assumes that when $p_m > 1/D$ (the SUERM link failure threshold), the Leaky Bucket counts up at a rate of $u(p_m - 1/D)$, where u is the rate at which MSUs are sent (this assumes that, due to retransmissions, the link is fully utilised by MSUs). The mean changeover time is then

$$T_{co} = T/(u(p_m - 1/D)) \qquad \text{(Eqn 3.10)}$$

where T is the SS7 Leaky bucket depth and D the count down rate. The mean changeover transient, Q_{co}, is then product of T_{co} and the net rate at which the transmit queue grows. This gives

$$Q_{co} = \left(\rho - \frac{(1-p_m)}{1+p_m uTL}\right)\left(\frac{T\overline{M}}{p_m - \frac{1}{D}}\right) \qquad \text{(Eqn 3.11)}$$

where ρ is the offered load and $\overline{M}$ the mean MSU length. This is solved (with respect to p_m) to find the maximum changeover transient.

3.4.2 Changeover Time

The previous subsection described a simple model for link changeover time vs error rate (which is also proposed in Q.703). This model implies however that the changeover time becomes infinite when p_m approaches $1/D$. As the link changeover time provides a measure of the changeover transient, an accurate procedure for determining link changeover time is desirable. Part of the work for this thesis has therefore been to obtain a more accurate model for link changeover time, which in turn required the Leaky Bucket to be modelled. Two methods were developed for this, which we now describe.

The first method was developed by I. Kerekes, from the Switched Networks Research Centre at the University of Wollongong [KER93]. This uses a Markov Chain to represent all possible states for the Leaky Bucket. For each Leaky Bucket level (i.e. from 1 to T), there are D possible values for the down counter (recall that the Leaky Bucket level is reduced by 1 when the down counter reaches D). With the SS7 values for T and D (64 and 256 respectively), the resulting Markov Chain has over 10000 states. While the mean changeover time may, in principle, be obtained from the resulting transition matrix, the large number of states makes this approach impractical. Instead an iterative approach is used. This starts with an estimate of the number of steps (i.e. Signal Unit) arrivals until changeover, assuming that the Leaky Bucket starts at 1 less than the changeover level. A recursive method is then used to give two expressions for N_{co},the number of steps to overflow the Leaky Bucket given that it starts empty (this is the desired quantity). The error term produced by the difference between these 2 values of N_{co} is used to get a new starting estimate. By repeating this process, the estimate of N_{co} becomes progressively more accurate. The second moment of N_{co} may be obtained similarly. Further details are in [KER93].

The second method was developed by the author, and forms one of the contributions of this thesis. In addition to providing the mean changeover time for the Leaky Bucket, this second approach is a general result for the mean first passage time for an M/G/1/K queue (i.e. the mean number of arrivals before the first overflow of an M/G/1/K queue, given that it starts with j customers ($j < K$)). Although other solutions for the first passage time exist [KUM75], they are impractical for large values of K. To maintain the broad treatment of this review, we present below an outline only of this technique. Further details are in appendix A.

The iterative method of [KER93] was developed to avoid solving a large (i.e. > 10000 state) Markov Chain. The second method reduces the number of states, by recognising that the Markov Chain formed by the Leaky Bucket has a renewal point (i.e. when the system becomes independent of past events) whenever the Leaky Bucket level goes down by one. This happens every D arrivals (assuming that the bucket does not overflow).

A Markov Chain may then be formed, where each state corresponds to a Leaky Bucket level. The transition probabilities are then determined by the probabilities of 0,1,...,n Signal Unit errors in D arrivals. As this Markov Chain is very similar to that used to model the M/G/1/K queue [GRO74], the results may be applied to the M/G/1/K case also. If the Leaky Bucket changes over at T, then a transition matrix P results, with dimensions of $T+1 \times T+1$. The Markov chain has an absorbing state at T (i.e. when the Leaky Bucket overflows). We form a sub-matrix Q by removing the T+1th row and column of P, and define N_{co} as the inverse of $I-Q$, where I is the identity matrix. The mean number of steps to reach the absorbing state (i.e. overflow the Leaky Bucket) is then the sum of the elements of the first row of N_{co} [KEM76]. A similar technique has been used independently in [KAN93.1] to model changeover times for the Errored Interval Monitor.

The problem of Leaky Bucket changeover times has also been addressed in [RAM93]. This work was prompted by a reported SUERM problem, which causes links to oscillate in and out of service (i.e. have multiple changeovers). This was also noted for Telecom Australia's SS7 links over radio bearers [SPA91], where the SUERM parameters were relaxed to reduce the number of spurious changeovers. [RAM93] also uses a Markov Chain approach to model Leaky Bucket changeover time, however his technique also provides the changeover time distribution.

3.4.3 Error Monitors for High Speed SS7 Links

The planned use of high speed (T1) links for SS7 signalling has led to a re-evaluation of SS7 performance monitors. The result has been a new scheme, proposed in [SCH94], known as the Errored Interval Monitor (EIM). This is to replace the SUERM on T1 (1.536 Mbps) SS7 links. The EIM was introduced in Chapter 2; here it is reviewed in more detail. We first outline the problems seen with the SUERM on high speed links, then consider how the EIM addresses these problems.

Problems with the SUERM on high speed links

[SCH94] proposes two criteria for SS7 monitors: that they should be both safe and effective. A safe monitor is one which removes a link from service after the error rate exceeds a maximum value, but before the transmit queue reaches a predetermined level Q_{max} This maximum error value is equivalent to the maximum sustainable error rate mentioned earlier. An effective error monitor is insensitive to short error bursts, i.e. those which increase the transmit queue to a level less than Q_{max}. These two criteria, safe and effective, result in three requirements for signalling error monitors, i.e. that they should:

1) Cause a link changeover when the maximum error rate is exceeded.

2) Tolerate burst errors for a specified period before initiating a changeover.

3) Ensure that the changeover transient (i.e. the amount of data queued when changeover is initiated) does not exceed a specified value.

[SCH94] rejects the SUERM for high speed links on the grounds that it "cannot be safe and effective". He uses the method of [SKO87] to obtain changeover transients for the SUERM, for both 56 kbps and 1.536Mbps links. His results show that while the SUERM restricts the changeover transient to around 3000 octets for a 56kbps link, a 1.536 Mbps link (with T scaled from 64 to 1755 in accordance with the increased link speed) has a changeover transient with an infinite peak at error probabilities close to the changeover threshold (i.e. 1/D). While this peak may be reduced by adjusting parameters, it remains of the order of megabytes (which translates to a delay of many seconds for a 1.536 Mbps link).

The reason for this increased transient may be seen from Equation 3.10 and Equation 3.11. In order to keep the same minimum changeover time as the 64 kbps SUERM at high error rates, the changeover level (T) must be scaled linearly, in proportion to the increase in link speed. However as the link speed increases, the service rate (u) increases also. This also increases the changeover transient (as shown by Equation 3.11), thereby accounting for the disproportionate increase noted in [SCH94].

The problem with the Leaky Bucket (i.e. SUERM) on high speed links is that by providing a bucket deep enough to ride over isolated error bursts (i.e. to satisfy requirement 2), T becomes so large that an unacceptable changeover transient occurs at error rates close to 1/D.

EIM Operation

To overcome the shortcomings of the SUERM on high speed links, [SCH94] proposes the EIM, which counts errored intervals instead of errored Signal Units (hence decoupling the parameter T form the link speed). The scheme operates by increasing a counter by U if an interval contains one or more Signal Unit errors. The counter is decremented by D for each error free interval, with a changeover occurring when the counter exceeds T (the proposed values for U, D and T are 144292, 9308 and 577169 respectively). We now consider how the EIM fulfils the three error monitoring requirements outlined above.

The parameters for the SS7 EIM are chosen so that a changeover will occur if the BER exceeds a value b_{max} (hence satisfying the first requirement). This is the maximum BER sustainable on an SS7 link, while still satisfying delay specifications. As mentioned previously, SS7 links transmit continuously, so that each EIM interval contains the same number of bits. Hence a given BER, b, results in an interval error probability p_I, where $p_I = 1 - (1-b)^N$ and N is the number of bits in an interval (this assumes independent errors). The maximum sustainable interval error probability p_{Imax} (corresponding to a BER of b_{max}), is then $1 - (1-b_{max})^N$. To ensure a changeover when p_{Imax} (and hence b_{max}) is exceeded, U and D are chosen so that $D/(U+D) = p_{Imax}$

A key aim of the EIM is to keep the changeover transient below a given value (i.e. the third requirement). The transmit queue (which determines the changeover transient) is estimated from the EIM counter value, multiplied by a scaling factor (this scaling factor is required so that a changeover occurs at b_{max}). The parameters U and D represent the respective estimated addition and reduction to this queue, due to errored and non-errored intervals. [SCH94] shows how to estimate the addition to the transmit queue (due to the delay introduced by the go-back-n protocol) when an error occurs. This is the basis for determining U. As the analysis is lengthy, we omit it here. Similar arguments are used to determine D and T, to satisfy a given objective for the changeover transient. The values chosen for U, D and T (144292, 9308 and 5771 respectively) are chosen to ensure that the changeover transient does not exceed 50000 octets [CHE93].

While placing an upper limit on the changeover transient, the values of U and D also guarantee a minimum changeover time of 4 intervals (i.e. 400 msec). Individual error bursts which are less than 400 msec will not cause changeovers (this is given as an objective in [CHE93]). Hence the SS7 EIM fulfils the three error monitoring requirements, i.e. to changeover when the maximum sustainable error rate is exceeded, to limit the changeover transient and to ride over short error bursts.

[KAN93.2] also proposes the EIM be used for high speed links. However he argues for the use of three sets of parameters, for short, medium and long links ([SCH94] suggests one set of parameters only). This is based on the observation of [HOU94], that the maximum sustainable BER changes with the link length. [CHE93] argues that multiple parameter sets are likely to cause operational difficulties (in particular when link lengths change, due to re-routing).

The changeover time versus error rate characteristic of the EIM is modelled in [KAN93.1], using a Markov Chain with an absorbing state (similar to the method developed by us). However the lack of memory associated with the EIM states make the problem far simpler than the Leaky Bucket analysis. The results of [KAN93.1] show that the current interval size (100 msec) produces a wide knee in the changeover time versus error rate curve. To sharpen this knee, i.e. to ensure a rapid changeover when the maximum BER is exceeded while minimising the changeover probability when the BER is less than the maximum sustainable value, [KAN93.2] proposes a shorter EIM interval.

3.5 Conclusions

This chapter has reviewed past work on SS7 performance modelling, Selective Repeat performance modelling, and SS7 error monitoring. This review has raised important issues regarding the performance modelling and error monitoring of ATM signalling links. These issues, which we summarise here, are addressed in the next two chapters.

As Selective Repeat has been chosen for ATM signalling, a delay performance model is needed. This should have the following features:

1) The increased queueing delay, due to a mixture of different MSU lengths, should be modelled (rather than assuming all MSUs to be the same size).

2) The effect of the change in error probabilities for MSUs of different lengths (within a given MSU length distribution) should be accounted for (rather than assuming the MSU error probability to be constant).

3) Delay percentiles should be provided, in addition to the mean delay.

4) The sensitivity of the mean delay (and delay percentiles), with respect to utilisation, is needed.

5) Variations in round trip delay (in particular between the first retransmission of a failed packet and a subsequent retransmission of that packet) should be modelled.

A new Selective Repeat delay performance model, which incorporates these features, is presented in Chapter 4.

While there has been much work on SS7 error monitoring, as seen by the review in this chapter, it has been directed towards current switching technologies, not ATM. Error monitoring for ATM signalling is discussed in draft recommendation Q.lm-nni [ITU94.1], which describes layer management of the Signalling AAL at the Network Network Interface (NNI). However no specific recommendations are made regarding the scheme to be used, except to say that it may be based on either the MSU error rate or performance information from the OAM layer (which monitors blocks of ATM cells). As a result, the choice of error monitor for ATM signalling links remains an open issue.

The possibility of using OAM Layer information, in addition to MSU error rates, gives rise to three potential schemes for ATM signalling error monitors (as identified in chapter 2). These schemes either:

1) count errored Signal Units (i.e. the SUERM).

2) count errored time intervals (i.e. the EIM).

3) monitor blocks of ATM cells, using the OAM capabilities.

These schemes are compared in Chapter 5 on the basis of how well they fulfil three requirements, i.e. that they:

1) cause a link changeover as soon as possible after the maximum error rate is exceeded.

2) tolerate burst errors for a specified period before initiating changeover.

3) ensure that the changeover transient (i.e. the amount of data queued when changeover is initiated) does not exceed a specified value.

4. ATM Signalling Performance Modelling

4.1 Introduction

This chapter examines the delay performance of ATM signalling links. We begin by outlining the Service Specific Connection Oriented Protocol (SSCOP), which is the link layer protocol intended for the Signalling AAL. This uses Selective Repeat error correction. We introduce a new model for Selective Repeat delay performance, which incorporate the features of the SSCOP. The results from this model, i.e. mean delay, delay percentiles and the sensitivity of each with respect to utilisation, are key inputs to the ATM error monitor design presented in chapter 5.

Section 4.2 describes SSCOP operation, focusing on the method for reporting errors. Section 4.3 presents a model for SSCOP mean delay performance, which is also a general method for calculating mean Selective Repeat delay. This model overcomes the deficiencies of previous Selective Repeat analyses noted in Chapter 3, and forms one of the major contributions of this thesis. Section 4.4 extends this model to give delay percentiles.

To test the assumptions underlying the new SSCOP delay model, Section 4.5 compares analytical delay results with those from a simulated SSCOP connection.

4.2 SSCOP

The SSCOP will be the first widely used implementation of a Selective Repeat protocol, and is intended for the signalling AAL (and other B-ISDN applications). The SSCOP employs a unique polling method to acknowledge packet arrivals and report errors. To place this method in context, we begin by describing previous Selective Repeat implementations.

Most studies of Selective Repeat performance are for idealised protocol models, where implementation details are not considered. Exceptions to this are [AHM89] and [NET90]. With the scheme of [AHM89], the receiver sends regular status packets to acknowledge packet arrivals and report errors. A similar scheme, i.e. regular status packets, is employed in [NET90], which considers the implementation of a transport layer using Selective Repeat over a high speed (i.e. 100 Mbps) link. If more than n status packets arrive without reporting the successful arrival of a previously sent packet, a retransmission is done. To avoid spurious retransmissions, the time for n status packets to arrive must exceed the round trip interval, which is measured during connection establishment.

If status packets are sent at regular intervals to report errors (and acknowledge successful arrivals), the time to report an error (and hence the delay increase) has 2 components: the round trip time and the time before the next status packet is sent. To eliminate this second component of delay, we proposed [EYE91] sending a status packet as soon as an error was detected, in addition to the regular status packet transmissions. This idea was also adopted (independently) in the SSCOP, whose operation we now describe.

The SSCOP uses dedicated poll and status packets to exchange state information. Poll packets are sent regularly by the SSCOP transmitter. When a poll arrives at the receiver, a status packet is returned immediately. This status packet, called a Solicited Status Response (or STAT), acknowledges successful packet arrivals, reports errored (i.e. missing) packets, and indicates how many additional packets the receiver is willing to accept. This latter function provides a sliding window flow control. In addition to the regular transmissions, a status packet is sent whenever a missing data packet is detected at the receiver (this is known as an Unsolicited Status Response (or USTAT)). As mentioned above, this reduces the time to report errors, and hence the resulting delay.

Each poll carries the sequence number of the most recently transmitted data packet. If this most recent packet is lost, the arrival of a poll will alert the receiver to this. To avoid measuring the round trip delay for the error detection scheme to work (as required in [NET90]), the SSCOP uses a scheme based on poll sequence numbers. Each poll carries a sequence number, which is returned in the responding status packet. As each data packet is sent, the sequence number of the most recent poll is recorded. If a status packet arrives which: a) carries a higher sequence number (i.e. for a subsequent poll) and b) does not acknowledge the packet arrival, then the packet is assumed lost and a retransmission is initiated. By using poll sequence numbers in this way, rather than measuring time (or the number of status packet arrivals in [NET90]), the SSCOP does not require the round trip interval to be known.

In addition to exchanging state information, poll and status packets are used to detect link failure, i.e. a "keep alive" function. A timer is maintained, which is reset whenever a poll or status packet arrives. Should this timer exceed the value contained in the variable Timer_NO-RESPONSE, the link is presumed to have failed, and the SSCOP connection is released.

The SSCOP error reporting scheme causes differences between the delay correcting an initial packet error and the delay correcting a failed retransmission. The delay correcting an initial error is the of order of the round trip time (i.e. twice the propagation delay). However should the retransmission fail, it is not recovered until one round trip after the next poll is sent. Hence the delay recovering a failed retransmission is determined by the interval between polls. This has a recommended value of 100 msec for signalling applications [ITU93.2]. Hence a signalling link with a round trip time less than 10 msec (typical within an urban area) will have (on average) an order of magnitude difference between the delay to recover an initial error and the delay to recover a failed retransmission. The need to account for this difference when modelling SSCOP delay is addressed in the next section.

4.3 SSCOP Delay Analysis

This section presents a new delay performance model of the SSCOP. The aim is to overcome the deficiencies of previous Selective Repeat analyses noted at the end of chapter 3. In particular, we model the effect of different packet sizes (both the change in queueing delay and packet error probability), and the effect of variations in the round trip delay inherent to the SSCOP. As the SSCOP uses a Selective Repeat ARQ, this model considers delay at both the transmitter and receiver. Transmitter delay is the time between a packet arrival at the transmit queue and the completion of a successful transmission. With the Selective Repeat ARQ, packets may also be delayed at the receiver (unlike go-back-n). This is to ensure correctly sequenced delivery of packets to higher layers. A packet arriving at the receiver will be buffered (and hence delayed) if any previous packets have not been successfully received. We begin by considering transmitter delay.

4.3.1 Transmitter Delay

The transmitter delay analysis assumes the following:

1) New packet arrivals are poisson.

2) The link (i.e. the ATM Virtual Channel Connection (VCC)) suffers a constant BER, where bit errors are independent (note: this is easily extended to an ATM cell error rate).

3) The link carries a range of packet sizes, where the error probability of each packet is a function of its length and the BER (previous analyses assume constant packet sizes and constant error probabilities).

4) Packets transmissions are not restricted by the SSCOP window mechanism (i.e. the SSCOP receiver credits).

5) The link (or VCC) is a Constant Bit Rate (CBR) connection [ITU.371]. This has a fixed bandwidth allocation, which is specified by a peak rate (which cannot be exceeded). We assume that all ATM cells are sent at the peak rate (e.g. 1 Mbps).

The mean transmitter delay for an SSCOP connection has two components, the queueing delay until the first transmission, and the virtual service time (i.e. the time required for a successful packet transmission). We first consider the queueing delay.

Queueing Delay

In addition to the terminology introduced in the previous chapter, we use the following:

λ_p poll arrival rate.

λ_r arrival rate for retransmitted packets

S_p, S_{p2} First two moments of poll service time.

S_r, S_{r2} First two moments of the service time for retransmitted packets

ρ_r Utilisation of the server by retransmitted packets

ρ_p Utilisation of the server due to the transmission of poll and status packets ($= 2\lambda_p S_p$). This assumes that poll and status packets arrive at the same rate, and have the same service time.

Packet arrivals, and poll and status arrivals, are assumed to be poisson (the arrival of poll and status packets will in fact be deterministic). Like [ANA86], we consider the arrival of negative acknowledgements (carried in status packets) to be independent of the past history of the system. The retransmitted packets, due to these negative acknowledgements, are assumed to form a separate poisson arrival process.

Retransmitted packets have top priority in the SSCOP protocol. Poll and status packets come next, with new packets having the lowest priority. We model the transmitter as a non-preemptive priority M/G/1 queue. The three arrival streams, retransmitted packets, poll and status packets, and new packets, have the priorities described above. Applying the standard result for M/G/1 non-preemptive priority queues ([JAI68]) gives the mean queueing delay for new packets, which is

$$Q_m = \frac{(2\lambda_p S_{p2} + \lambda_r S_{r2} + \lambda S_{m2})}{2(1-(\rho_p+\rho_r))(1-(\rho_p+\rho_r+\rho_m))} \qquad \text{(Eqn 4.1)}$$

To evaluate Q_m, we need expressions for S_{r2}, S_r and λ_r (the latter two give ρ_r). To get these, we consider the probability p_{ri} that a retransmitted packet has a length l_i (and hence a service time S_i). This is

$$p_{ri} = \frac{(m_i p_i)/(1-p_i)}{\sum_{j=1}^{h} (m_j p_j)/(1-p_j)} \qquad \text{(Eqn 4.2)}$$

where m_i the probability that an MSU has a length l_i bits. Hence the first two moments of the service time for retransmitted packets are $S_r = \sum_{i=1}^{h} p_{ri} S_i$ and $S_{r2} = \sum_{i=1}^{h} p_{ri} S_i^2$. The arrival rate of retransmissions is

$$\lambda_r = \sum_{i=1}^{h} (\lambda m_i p_i) / (1 - p_i)$$

so that the utilisation of the server by errored packets is $\rho_r = \lambda_r S_r$. By incorporating the first 2 moments of service time, the new expression for Selective Repeat (and SSCOP) transmitter delay given in Equation 4.1 incorporates the increased queueing delay due to variations in packet size (unlike previous Selective Repeat analyses)[1].

Virtual Service Time

The virtual service time is the time to successfully transmit a packet. We use the same notation for this, i.e. S_v, as in the go-back-n analyses of chapter 3 (the difference is that for go-back-n, S_v is also the time that the packet effectively occupies the server). To determine S_v, we consider the retransmission delay introduced by the SSCOP. Figure 4.1 shows the SSCOP retransmission scheme. When a packet error is first detected (at A in the figure), an Unsolicited Status packet is sent immediately, which causes a retransmission to arrive at B. The time to correct this first error (for a packet of length l_i) is $T_L + S_i + S_s + T_d$.

T_d is the delay for a packet error to be detected at the receiver. As this detection occurs when the next packet (or poll) arrives, T_d has a mean value of $\overline{T_d} = 1/(\lambda + \lambda_p)$.

S_s represents additional delay, which occurs if a packet is already in the server when a negative acknowledgement arrives (the transmission of the packet in the server must complete before the errored packet can be re-sent). The mean value of this additional delay is given as

$$S_s = \sum_{i=1}^{h} (\textit{utilisation of server by packets of length } l_i) \qquad \text{(Eqn 4.3)}$$

$$\times\ (\textit{mean remaining service time for packets of length } l_i)$$

1. The use of the non-preemptive priority M/G/1 queue to model Selective Repeat transmitter delay was first proposed by us in [EYE91.2]

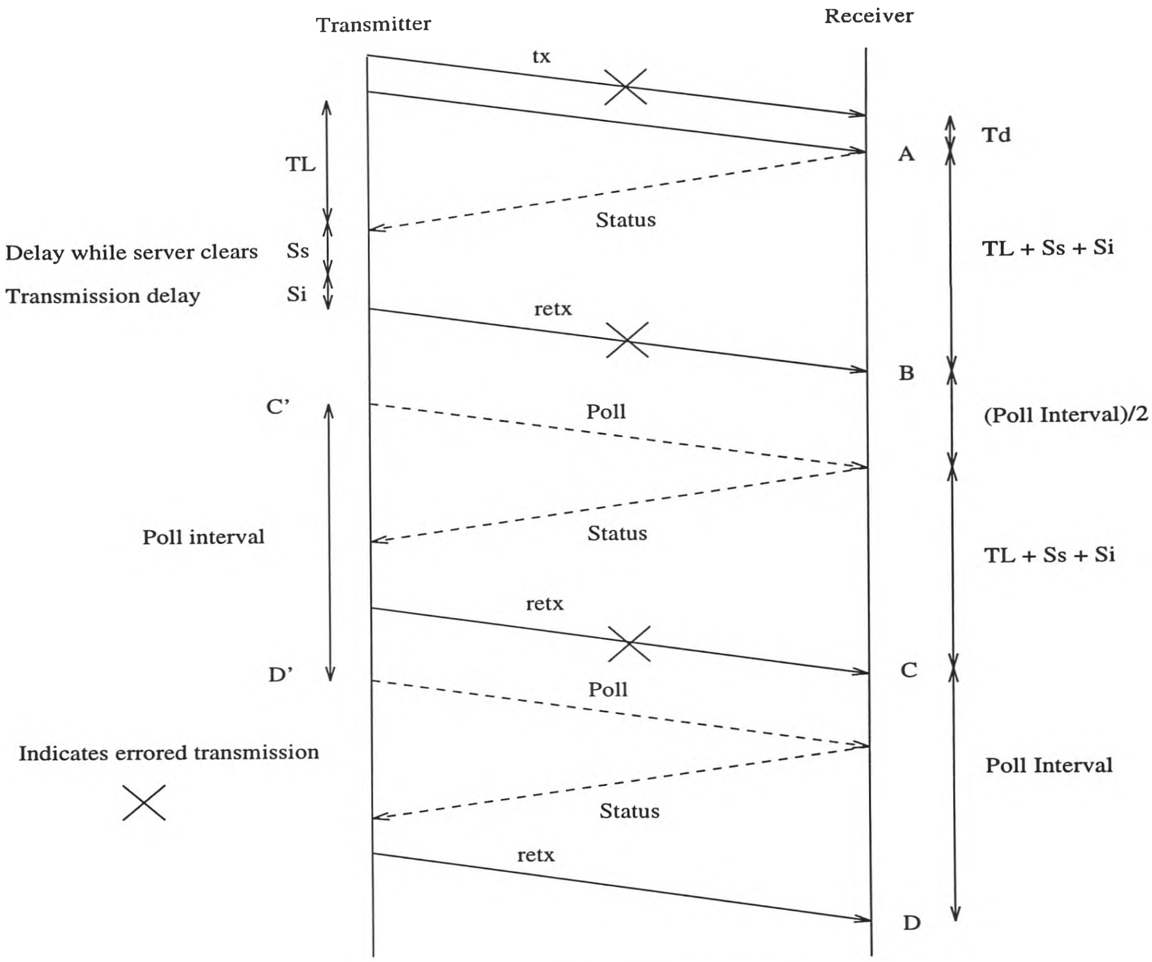


Figure 4.1 SSCOP retransmission scheme

$$= \sum_{i=1}^{h} \lambda m_i S_i^2/2$$

The retransmission arriving at B, which we assume to fail as well, is not recovered until a poll packet arrives at the receiver (the SSCOP transmitter sends polls at regular intervals). This prompts a Solicited Status packet, which reports the failed retransmission. Hence the mean time required for the second retransmission (which arrives at C) is the mean interval until the next poll (which we take to be $1/(2\lambda_p)$), plus $T_L + S_i + \overline{S_s}$, as shown in the figure ($\overline{S_s}$. is the mean value of S_s). If the poll interval (i.e $1/\lambda_p$) exceeds $T_L + S_i + \overline{S_s}$, the additional time for third (and all subsequent) retransmissions is just the Poll interval. This can be seen from the figure by noting that the time between C and D is the same as the time between C' and D', which is also the poll interval.

The virtual service time for a packet of length l_i, as a function of the number of retransmissions, is given by

$$
\begin{aligned}
S_{oi} &= S_i && \textit{for 0 retransmissions} \\
S_{1i} &= S_{oi} + T_L + S_i + \overline{S_s} + \overline{T_d} && \textit{for 1 retransmission} \\
S_{2i} &= S_{1i} + T_L + S_i + \overline{S_s} + 1/(2\lambda_p) && \textit{for 2 retransmisions} \\
&\dots \\
S_{ni} &= S_{(n-1)i} + 1/\lambda_p && \textit{for n retransmissions } n > 2
\end{aligned}
\quad \text{(Eqn 4.4)}
$$

(Note: if the Poll interval is less than $T_L + S_i + \overline{S_s}$, then $S_{ni} = S_{(n-1)i} + T_L + S_i + \overline{S_s} + 1/2\lambda_p \quad n > 2$)

Using Equation 4.4 we get the overall mean virtual service time, i.e. averaged over all packet lengths. This is

$$S_v = \sum_{i=1}^{h} m_i \sum_{n=0}^{\infty} p_i^n (1 - p_i) S_{ni} \quad \text{(Eqn 4.5)}$$

Equation 4.4 and Equation 4.5 capture the effect of variations in round trip delay and packet size, which is the aim of this new analysis. The total mean transmitter delay, D_t, is the sum of Q_m and S_v.

4.3.2 Receiver Delay

Here we consider the mean Selective Repeat (and hence SSCOP) receiver delay.[2] In addition to the assumptions for the transmitter delay calculation, we also assume that new packets (i.e. not retransmissions) arrive at the receiver as a poisson process, with rate λ. Packets with errors on their first transmission attempt also arrive as a Poisson process.

To model receiver delay, we consider the packets which suffer an error on their first transmission attempt. As these errored arrivals are assumed pois-

2. An earlier version of this analysis, which assumed constant packet error probabilities, appeared in [EYE94.1]

son, then the error processing at the receiver can be modelled as an M/G/∞ queue. The "customers" for this queue are the errored packets, with the service time being the time to correct an errored packet. We now consider an arbitrary packet arrival at the receiver. If we assume this packet to be error free (but not a retransmission of an earlier error), then it will not influence the M/G/∞ queue. However the packet will remain in the receiver buffer (and thus be delayed) until all errors (i.e. customers) currently in the M/G/∞ queue have been corrected. The delay for this arbitrary packet will therefore equal the largest remaining service time (i.e. remaining time until correction) of all the customers (i.e. errored packets) currently in the queue. However the delay for an arbitrary packet is also the delay seen by poisson arrivals [GRO74]. Hence the largest remaining service time (which we will call the maximum residual service time) is the delay seen by new error free packets (i.e. not retransmissions) which arrive at the receiver, as we assume these arrivals to be poisson. As it is this delay that we seek, the task is to find the steady state distribution (and the moments) of the maximum residual service time for this M/G/∞ queue.

This problem (i.e. distribution of the maximum residual service time for M/G/∞ queues) is partly solved in [BRO69], who gives an equivalent result for a transient case (although he does not consider it in the context of the maximum residual service time). [BRO69] starts by considering a single customer in an M/G/∞ queue, and arrives at the following[3]

$$F(x) = p\,(customer\ arriving\ during\ (0,t)\ is\ served\ by\ t+x)$$
$$= \frac{1}{\lambda t}\int_0^t B(t+x-y)\,\lambda dy \qquad \text{(Eqn 4.6)}$$

where $B(t)$ is the customer service time distribution. [BRO69] then extends this to all customers arriving at the M/G/∞ queue, to give

3. [BRO69] considers a more general case, with batch arrivals. In accordance with our model, we have adjusted Equation 4.6 to give the distribution for single arrivals.

$$p\,(all\ customers\ arriving\ in\ (0,t)\ are\ served\ by\ t+x) \qquad \text{(Eqn 4.7)}$$
$$\begin{aligned} &= \sum_{n=0}^{\infty} \frac{(\lambda t)^n e^{-\lambda t}}{n!} F^n(x) \\ &= \sum_{n=0}^{\infty} \frac{e^{-\lambda t}}{n!} \left(\int_0^t B(t+x-y)\,\lambda dy \right)^n \\ &= e^{-\lambda t} e^{\int_0^t B(t+x-y)\,\lambda dy} \end{aligned}$$

(note: in presenting the steps of Equation 4.7, we have corrected a small error in [BRO69]). If we extend this transient result to the steady state (i.e. $t = \infty$), then we get the distribution of the time to complete service for all the customers in an M/G/∞ queue at an arbitrary instant. In other words, the steady state form of the above expression gives the residual service time distribution, which we seek. To this end, we note that Equation 4.7 can be written as

$$e^{-\lambda\left(t - \int_x^{(t+x)} B(z)\,dz \right)} \qquad \text{(Eqn 4.8)}$$

and that

$$\begin{aligned} & t - \int_x^{t+x} B(z)\,dz \qquad \text{(Eqn 4.9)} \\ &= \int_0^{t+x} (1 - B(z))\,dz - \int_0^x (1 - B(z))\,dz \end{aligned}$$

Setting $t = \infty$, this becomes

$$B - \int_0^x (1 - B(z))\,dz \qquad \text{(Eqn 4.10)}$$

where B is the mean service time. Hence the distribution of the maximum residual service time for an M/G/∞ queue (i.e. the steady state form of the expression in [BRO69]) is

$$
\begin{aligned}
F(x) &= e^{-\lambda\left(B - \int_0^x (1 - B(z))\, dz\right)} \\
&= e^{-\rho} e^{\lambda \int_0^x (1 - B(z))\, dz}
\end{aligned}
\qquad \text{(Eqn 4.11)}
$$

As outlined earlier, this gives the distribution of the Selective Repeat receiver delay, where $B(t)$ is the distribution of the time for an errored packet to be corrected. λ in Equation 4.11 becomes λ_e (for the purposes of the Selective Repeat receiver delay calculation), where λ_e is the rate at which packets with errors on their first transmission arrive at the receiver. Similarly ρ becomes ρ_e (which will be derived subsequently). In the context of this thesis, Equation 4.11 is an important result. In particular, $B(t)$ can be written to incorporate variations in round trip delay and packet error probability (as will be shown), thereby achieving the aim of this new Selective Repeat (and SSCOP) analysis. Equation 4.11 has also been obtained independently by [GOU94][4], using a different approach. However [GOU94] (who also considers the delay for the SSCOP), leaves it in this form (i.e. the delay distribution only). We now extend this result to give the mean Selective Repeat receiver delay.

To determine this mean delay, we first obtain expressions for $B(t)$ and hence $\int_0^t (1 - B(z))\, dz$. To do this, we define $D_1, D_2, \ldots, D_n$ as the mean correction times for packets requiring $1, 2, \ldots, n$ retransmission attempts. Following the approach for the transmitter delay, we get

$$
\begin{aligned}
D_1 &= T_L + S_e + \overline{S_s} \qquad \textit{for 1 retransmission} \\
D_2 &= D_1 + T_L + S_e + \overline{S_s} + 1/(2\lambda_p) \qquad \textit{for 2 retransmissions} \\
D_3 &= D_2 + 1/\lambda_p \qquad \textit{for 3 retransmissions} \\
&\ldots \\
D_n &= D_{n-1} + 1/\lambda_p \qquad n > 2
\end{aligned}
\qquad \text{(Eqn 4.12)}
$$

4. The results (but not the derivation) were also reported in [HEB94]

Here we do not include T_d, the interval before an initial error is detected, because receiver delay (by definition) does not start until this detection occurs. S_e is the mean service time for errored packets[5], which is $\sum_{i=1}^{h} m_{ei} S_i$ Here m_{ei} is the probability that an packet arriving at the receiver with an error in its **first** transmission has length l_i. This probability is

$$m_{ei} = \frac{m_i p_i}{\sum_{j=1}^{h} m_j p_j} \qquad \text{(Eqn 4.13)}$$

(note: m_{ei} is not the same as p_{ri} in Equation 4.2, which is the probability that ***any*** retransmitted packet has length l_i). The mean correction delay (D_e) for errored packets (i.e. the interval between the initial error being detected at the receiver and a correct version of the packet arriving) is then

$$D_e = \sum_{i=1}^{h} m_{ei} \sum_{j=0}^{\infty} p_i^j (1 - p_i) D_{j+1} \qquad \text{(Eqn 4.14)}$$

so that the utilisation of the M/G/∞ queue used by the receiver delay model is $\rho_e = \lambda_e D_e$. λ_e is the arrival rate, at the receiver, of packets with errors on their first transmission attempt. This is given by

$$\lambda_e = \lambda \sum_{i=1}^{h} m_i p_i \qquad \text{(Eqn 4.15)}$$

Having defined $D_1, D_2, \ldots, D_n$, we now write $B(t)$ as follows

5. Note that Equation 4.4, which lists similar delays as seen at the SSCOP transmitter, presents these delays as a function of the packet length. To allow tractability for receiver delay calculation which follows, Equation 4.12 and Equation 4.14 consider the mean delay only. Strictly speaking, the value of S_e on the first line of Equation 4.12 will be different from its value on the second line, as longer packets are more likely to suffer 2 (or more) retransmissions. Once again, in the interest of tractability, we assume S_e to be constant.

$$B(t) = p(\textit{correction time} \le t) \qquad \text{(Eqn 4.16)}$$

$$= 0 \qquad t < D_1$$

$$= \sum_{i=1}^{h} m_{ei}(1-p_i) \qquad D_1 \le t < D_2$$

$$= \sum_{i=1}^{h} m_{ei}(1-p_i) + \sum_{i=1}^{h} m_{ei}p_i(1-p_i) \qquad D_2 \le t < D_3$$

$$\ldots$$

$$= \sum_{j=0}^{n-2} \sum_{i=0}^{h} m_{ei}p_i^j(1-p_i) \qquad D_{n-1} \le t < D_n, n \ge 2$$

(Equation 4.16 implies that the $D_1, D_2, \ldots, D_n$ terms are deterministic. In practice they will not be, due to the different transmission time for different packet lengths). To obtain the Selective Repeat receiver delay distribution in Equation 4.11, we need $\int_0^t (1 - B(z))\,dz$. Using Equation 4.16, we get (after some manipulation)

$$F_r(t) = e^{-\rho_e} e^{\lambda_e \int_0^t (1 - B(z))\,dz} \qquad \text{(Eqn 4.17)}$$

$$= e^{-\rho_e} e^{\lambda_e t} \qquad 0 \le t \le D_1$$

$$= e^{-\rho_e} e^{\lambda_e (P_{(n-1)} t + Q_{(n-1)})} \qquad D_n \le t < D_{n+1}, n \ge 1$$

where

$$P_n = 1 - \sum_{j=0}^{n} \sum_{i=1}^{h} m_{ei}p_i^j(1-p_i) \qquad \text{(Eqn 4.18)}$$

$$Q_n = \sum_{j=0}^{n} \sum_{i=1}^{h} m_{ei}p_i^j(1-p_i)\,D_{j+1}$$

Equation 4.17 and Equation 4.18 together give the distribution of the SSCOP receiver delay in terms of the parameters of the SSCOP connection (Equation 4.11 does not). In particular, this distribution accounts for both the variation of the round trip time inherent to the SSCOP and the variation in packet error probability due to different packet sizes. The incorporation of these two effects is a key aim of this analysis.

We now evaluate the distribution in Equation 4.17 to get the mean delay, i.e. $\int_0^\infty t dF_r(t)$. To do this, we evaluate the integral piecewise, i.e. from 0 to D_1, D_1 to D_2 and so on. This gives the mean SSCOP (and Selective Repeat) receiver delay, D_r, which is

$$D_r = \lambda_e e^{-\rho_e} \int_0^{D_1} t e^{\lambda_e t} dt \qquad \text{(Eqn 4.19)}$$

$$+ \sum_{n=1}^{\infty} \lambda_e P_{n-1} e^{\lambda_e Q_{n-1}} e^{-\rho_e} \int_{D_n}^{D_{n+1}} t e^{\lambda_e P_{n-1} t} dt$$

$$= e^{-\rho_e} \Bigg[e^{\lambda_e D_1} (D_1 - 1/\lambda_e) + 1/\lambda_e + \sum_{n=1}^{\infty} e^{\lambda_e Q_{n-1}}$$

$$\times \Big(e^{\lambda_e P_{n-1} D_{n+1}} (D_{n+1} - 1/(\lambda_e P_{n-1}))$$

$$- e^{\lambda_e P_{n-1} D_n} (D_n - 1/(\lambda_e P_{n-1})) \Big) \Bigg]$$

While Equation 4.19 contains an infinite series, in practice very accurate results for the mean receiver delay may be obtained by evaluating the first few terms. The total mean delay for the SSCOP D_{tot}, i.e. from the time a packet joins the transmit queue until it is released from the receiver buffer (but not counting propagation delay) is then $D_{tot} = D_t + D_r$

4.4 SSCOP Delay Percentiles

As recommendation E.733 gives objectives for delay percentiles, in addition to mean delay, a method for determining these is needed. We outline a basic procedure for this, then present some refinements to it.

4.4.1 Basic Procedure

To determine SSCOP delay percentiles, we require the total SSCOP delay distribution (i.e. transmitter plus receiver delay). We begin by considering these two distributions separately. The receiver delay distribution has been obtained in the previous section (i.e. Equation 4.17). We approximate the transmitter delay distribution by ignoring the queueing delay and initial transmission time, and considering only the retransmission delay (i.e. from when the packet first leaves the server until a successful transmission is completed). We denote this retransmission delay as D_a (it will be 0 if the initial transmission succeeds). The error due to this approximation will be tested in the next section by comparing analytical and simulation results.

We assume D_a to take on discrete values, which are determined by the number of retransmissions. Similar to Equation 4.4, we express D_a as a function of the number of transmissions, which gives

$$\begin{aligned}
&D_{a0} = 0 \qquad \textit{for } 0 \textit{ retransmissions} && \text{(Eqn 4.20)}\\
&D_{a1} = S_m + T_L + \overline{S_s} + \overline{T_d} \qquad \textit{for } 1 \textit{ retransmission}\\
&D_{a2} = D_{a1} + T_L + S_m + \overline{S_s} + 1/(2\lambda_p) \qquad \textit{for } 2 \textit{ retransmissions}\\
&\ldots\\
&D_{an} = D_{an-1} + 1/\lambda_p \qquad \textit{for } n \textit{ retransmissions}, n > 2
\end{aligned}$$

In addition to omitting the time for the initial transmission, there is an additional difference between the terms in Equation 4.20 and Equation 4.4. Equation 4.4 gives the packet service time (S_i) as a function of the packet length. To maintain tractability however, the service time in Equation 4.20 (S_m) is the mean over all packet lengths, similar to Equation 4.12. (note: as is the case for the S_e term in Equation 4.12, the S_m values in Equation 4.20 will, in practice, vary with the number of retransmissions. Here we assume that S_m remains constant).

Using the terms in Equation 4.20, the distribution of D_a is then

$$
\begin{aligned}
F_{D_a}(t) &= p\,(retransmission\ delay \le t) \\
&= \sum_{i=1}^{h} m_i(1-p_i) \qquad t = 0 \\
&= \sum_{j=0}^{1}\sum_{i=1}^{h} m_i p_i^j(1-p_i) \qquad t \le D_{a1} \\
&= \sum_{j=0}^{2}\sum_{i=1}^{h} m_i p_i^j(1-p_i) \qquad t \le D_{a2} \\
&\ldots \\
&= \sum_{j=0}^{n}\sum_{i=1}^{h} m_i p_i^j(1-p_i) \qquad t \le D_{an}
\end{aligned}
\tag{Eqn 4.21}
$$

This is simply the probability that the number of packet retransmissions is $\le n$ (averaged over the set of packet error probabilities (i.e. $i = 1\ldots h$)). We combine this approximate transmitter delay distribution (i.e. $F_{D_a}(t)$) with the receiver delay distribution in Equation 4.17 ($F_r(t)$) to give the overall SSCOP delay distribution. By assuming $F_{D_a}(t)$ and $F_r(t)$ to be independent, we get

$$F_s(t) = F_{D_a}(t)\,F_r(t) \tag{Eqn 4.22}$$

To explain Equation 4.22, we consider an instant (T') at which a test packet arrives at the SSCOP receiver. Regardless of whether the test packet is errored or not, the receiver delay distribution seen at T' (by the test packet) is given by $F_r(t)$. Similarly the retransmission delay distribution at T' (which approximates the transmitter delay distribution) is $F_{D_a}(t)$. Hence the SSCOP delay seen by the test packet is $max\,(retransmission\ delay\ after\ T', receiver\ delay\ after\ T')$. The distribution of this is

$$
\begin{aligned}
&p\,(retransmission\ delay \le t\ and\ receiver\ delay \le t) \\
&= F_{D_a}(t)\,F_r(t)
\end{aligned}
\tag{Eqn 4.23}
$$

i.e. the same as Equation 4.22. The idea of expressing the entire SSCOP delay distribution as the product shown in Equation 4.22 first appeared in [GOU94]. However, as mentioned, the general expression for the receiver delay distribution (Equation 4.11) was developed independently by [GOU94] and us. Equation 4.17 and Equation 4.18, which give the receiver delay distribution in terms of the SSCOP parameters, as well as the closed form expression for the mean receiver delay (Equation 4.19) have not appeared previously (to our knowledge).

To obtain SSCOP delay percentiles (our overall aim) we use an iterative method, based on the SSCOP delay distribution. This begins with an estimate (t_p') of the desired n^{th} delay percentile, t_p. Equation 4.22 is then used to determine the actual percentile corresponding to t_p'. This is used to make a better estimate of t_p', with the process continuing until the estimate is within a specified accuracy (e.g. 0.01%) of the desired percentile.

4.4.2 A More Accurate Approach

Here we describe refinements to the procedures described previously, which give a more accurate result for the receiver delay percentiles.

The mean receiver delay calculation in section 4.3.2 assumes poisson arrivals at the receiver. Simulation results in section 4.5 show that the inaccuracies due to this assumption can be ignored, except at low error rates, where the mean delay is small (less than 1 msec for our examples). However assuming poisson arrivals causes the receiver delay percentiles to be underestimated, by as much as 50% in some instances. This is due to the operation of the SSCOP, where the packet which arrives after an error (and which causes the error to be detected) is delayed for the entire correction time (under the poisson arrival assumption, the probability that a packet will arrive immediately after an error, and be delayed for the entire correction time (instead of a part thereof), is small)

To overcome this inaccuracy in the receiver delay distribution, we consider the error free packets arriving at the receiver to form two groups, those which come immediately after an error (and which detect the error) and those which arrive otherwise. The probability that a packet belongs to the first group is $p_{first} = \sum_{i=1}^{h} m_i p_i \sum_{j=1}^{h} m_j (1 - p_j)$, with the probability of belonging to the second group being $p_{second} = 1 - p_{first}$.

We then form a second (i.e. revised) version of the complementary receiver delay distribution, which we denote $F'_{r2}(t)$. This is given as

$$\begin{aligned} F'_{r2}(t) &= p(delay > t) \\ &= p(delay > t/first)\, p_{first} + p(delay > t/second)\, p_{second} \\ &= p_{corr}(t)\, p_{first} + F'_r(t)\, p_{second} \end{aligned} \qquad \text{(Eqn 4.24)}$$

where $F'_r(t) = 1 - F_r(t)$ and $p_{corr}(t)$ is the complementary distribution of the retransmission delay, This is

$$\begin{aligned} p_{corr}(t) &= 1 \qquad t \le D_1 \\ &= 1 - \sum_{i=1}^{h} m_i (1 - p_i) \qquad D_1 < t \le D_2 \\ &= 1 - \sum_{i=1}^{h} m_i ((1 - p_i) + p_i (1 - p_i)) \qquad D_2 < t \le D_3 \\ &\ldots \\ &= 1 - \sum_{i=1}^{h} m_i \sum_{j=0}^{n-1} p_i^j (1 - p_i) \qquad D_n < t \le D_{n+1} \end{aligned} \qquad \text{(Eqn 4.25)}$$

i.e. the probability that a packet will require at least 0, 1 ... n+1 retransmission attempts.

To further improve the accuracy of the receiver delay distribution, we revise another simplifying assumption, namely that the correction time for a packet requiring n retransmissions has a fixed value D_n (this allowed the closed form expression for the mean receiver delay in Equation 4.19). While simulation results show that this assumption causes negligible errors for the mean delay (as will be seen in the next section), this is not so for percentile calculations. To model the variability in these correction times (and hence improve the percentile calculation accuracy), we introduce a random variable T_{poll}, which is the delay between an error and the next poll arrival (previously we assumed a fixed value of $1/(2\lambda_p)$ (the mean). T_{poll} replaces $1/(2\lambda_p)$ on the second line of Equation 4.12.

(While this change models the variation in delay for poll arrivals, fixed (i.e. mean) values are still assumed (to maintain tractability) for both the service time (i.e. S_e) and the delay to clear the server (i.e. $\bar{S}$)).

In addition to accounting for the variation of the delay for poll arrivals, we also account for variations in the time between an errored packet arriving at the SSCOP receiver and its subsequent detection. Similar to the correction time case, we introduce a random variable T_{detect}, which replaces $\overline{T_d}$ in Equation 4.20.

Introducing the random variables T_{poll} and T_{detect} changes the receiver delay distribution $F'_{r2}(t)$ to a conditional distribution, i.e. $F'_{r2}(t/T_{poll} = x, T_{detect} = y)$ We assume T_{poll}, which lies between 0 and the Poll interval, to be uniformly distributed (in the absence of other statistics regarding T_{poll}, the uniform distribution seems the best choice). T_{detect} has the same distribution as the time between packet arrivals at the receiver. We have previously assumed this distribution to be exponential. The overall delay distribution is determined numerically, by averaging $F'_{r2}(t/T_{poll} = x, T_{detect} = y)$ over the range of values for T_{poll} and T_{detect}.

4.4.3 Sensitivity

Recommendation E.733 also gives specifications for the sensitivity, with respect to utilisation, of the mean delay and the 99th delay percentile. Hence expressions for these sensitivities are required for ATM signalling links (and accordingly for the SSCOP). [SKO87] provides an expression for SS7 delay sensitivity with respect to utilisation, which is obtained easily from the ITU SS7 delay formulae (which express mean delay as a function of the utilisation).

As the delay expressions developed here cannot be not expressed as a function of the utilisation (unlike the SS7 delay formulae), closed form expressions for these sensitivities are not available. We therefore generate the required sensitivities numerically, i.e. by changing the utilisation up and down (by 1%) around the desired operating point and measuring the resulting gradient. This approach, which is essentially a graphical technique, is also suggested by E.733.

4.5 SSCOP Delay Results

This section presents SSCOP delay results arising from the previous analyses. The results are compared with those from a simulated SSCOP link.

4.5.1 Simulation Overview

The SSCOP delay analysis relies on three major simplifying assumptions. These are:

1) Packet arrivals at the SSCOP **receiver** are poisson. This implies that the output process of the SSCOP transmitter is also poisson.

2) Packets retransmissions arrive at the transmit queue as a poisson process.

3) The correction time for a packet requiring n retransmissions has a fixed value D_n.

To test the accuracy of the analysis in the light of the above assumptions, we have developed a discrete event simulation of an SSCOP link, which relaxes the assumptions. This simulation, which has been written in C and runs on a

SUN Sparcstation, mirrors the SSCOP protocol described in chapter 2. In particular:

- No assumptions are made about either the arrival process of packets at the receiver, or of retransmitted packets at the transmitter. Any correlations in these arrivals are captured accurately (to the resolution of the timer maintained by the simulation, i.e. 1 usec).

- The history of each packet (e.g. arrival time, length, transmission time, if retransmitted etc.) is kept until the packet is released from the system (i.e. when all prior packets have been acknowledged). At this time the following statistics are recorded: queueing delay, virtual service time, receiver delay and the total delay (i.e. the sum of the previous 3 terms).

- The SSCOP packet acknowledgement/error reporting processes are accurately modelled.

- A systematic procedure is used to determine confidence intervals for mean delays and delay percentiles. In general, the simulation continues running until the width of the 95% confidence interval is less than 5% of the measured delay (or delay percentile).

However the simulation still assumes poisson arrivals for new packets at the transmitter. In addition, independent errors are assumed, where the error probability of each packet is a function of the packet length and the bit error rate (the packet length distribution and the bit error rate are both simulation parameters). The simulation considers packets only, and does not consider the effect of segmenting these packets into ATM cells. Also the simulation does not incorporate processing delays at the SSCOP transmitter or receiver.

In addition to validating the overall SSCOP modelling approach (in particular the queueing models used for transmitter and receiver delay), the simulation results have prompted the refinements to the delay percentile calculation given in section 4.4.2

4.5.2 Results

We consider the delay performance of the SSCOP, as seen by a typical ATM signalling VCC. The VCC peak rate is 1 Mbps, with equal proportions of 2 cell and 10 cell packets (or Message Signal Units (MSUs)). These MSU sizes correspond approximately to those expected for B-ISDN CS-2 signalling messages [ITU93]. The mean arrival rate is 157 MSUs/sec, giving a utilisation of 0.4. The round trip time is 5 msec, with a poll interval of 100 msec, as recommended in [ITU93.2]. Where applicable, results are compared with those from the literature.

The simulated SSCOP results presented here are the points marked by a '+' sign. Although no error bars are shown for these simulated data points, the 95% confidence interval width is less than 5% of the simulated percentile values in most cases (exceptions to this are the points at low error rates, where the delay increase due to errors is negligible. Under these circumstances it is difficult to achieve narrow confidence intervals, due to the large variance in the delay. However as the delay results in this region are of little interest, the wider confidence intervals may be ignored).

Figure 4.2 shows the results for the mean SSCOP delay, at both the transmitter and the receiver. Figure 4.3 shows delay percentiles. We begin with the mean delay results

Figure 4.2a shows the mean transmitter queueing delay (i.e. delay until the first transmission) versus bit error rate. The upper plot is for the described mix of MSU sizes, while the bottom plot keeps all MSUs at 6 cells (the mean length). The individual data points are from the simulation. As seen, assuming constant MSU sizes (as required by previous analyses), rather than accounting for the actual packet size distribution, causes the mean transmitter queueing delay to be under-estimated. The increased delay when the message size variance is considered is due to

a) The effect of the message size variance on the queueing behaviour.

b) The increased utilisation of the transmitter queue, due to retransmissions of long messages (which have a higher error probability with our revised model).

In addition, the simulation results in Figure 4.2a show that modelling retransmitted packets as poisson arrivals introduces no appreciable error. This was also observed in [ANA86].

Figure 4.2b shows the results when the receiver delay method of [ROS89] (which assumes constant round trip intervals) is used to model SSCOP receiver delay. The individual data points show the simulated receiver delay. As the method of [ROS89] assumes a constant packet size, the simulation also uses fixed length packets (6 ATM cells), to show more clearly the effect of assuming constant round trip times. An immediate problem with the method of [ROS89] is to choose an appropriate value for the round trip time. The lower curve shows receiver delay with a round trip time of twice the propagation delay plus message transmission time (i.e. $T_L + S_m$), while the upper curve is for a round trip time equal to the Poll interval (these represent lower and upper bounds (respectively) for the SSCOP round trip time). The middle curve sets the round trip delay to the expected value (the jagged edges are due to quantisation errors when setting the number of packets in one round trip to an integral value, as required by the analysis). As shown, the assumption of a fixed round trip delay gives inaccurate results when the round trip interval varies, the case with the SSCOP.

Figure 4.2c shows the receiver delay obtained using the new method, i.e. Equation 4.19. To allow comparison with Figure 4.2b, constant MSU sizes are again used. By modelling the variations in round trip delay (due to the operation of the SSCOP), a much closer match with the simulation results is seen. Figure 4.2b and Figure 4.2c show delay as a function of the packet error probability, rather than the bit error rate. Again this is to allow comparison with the results of [ROS89], which give receiver delay as a function of packet error probability.

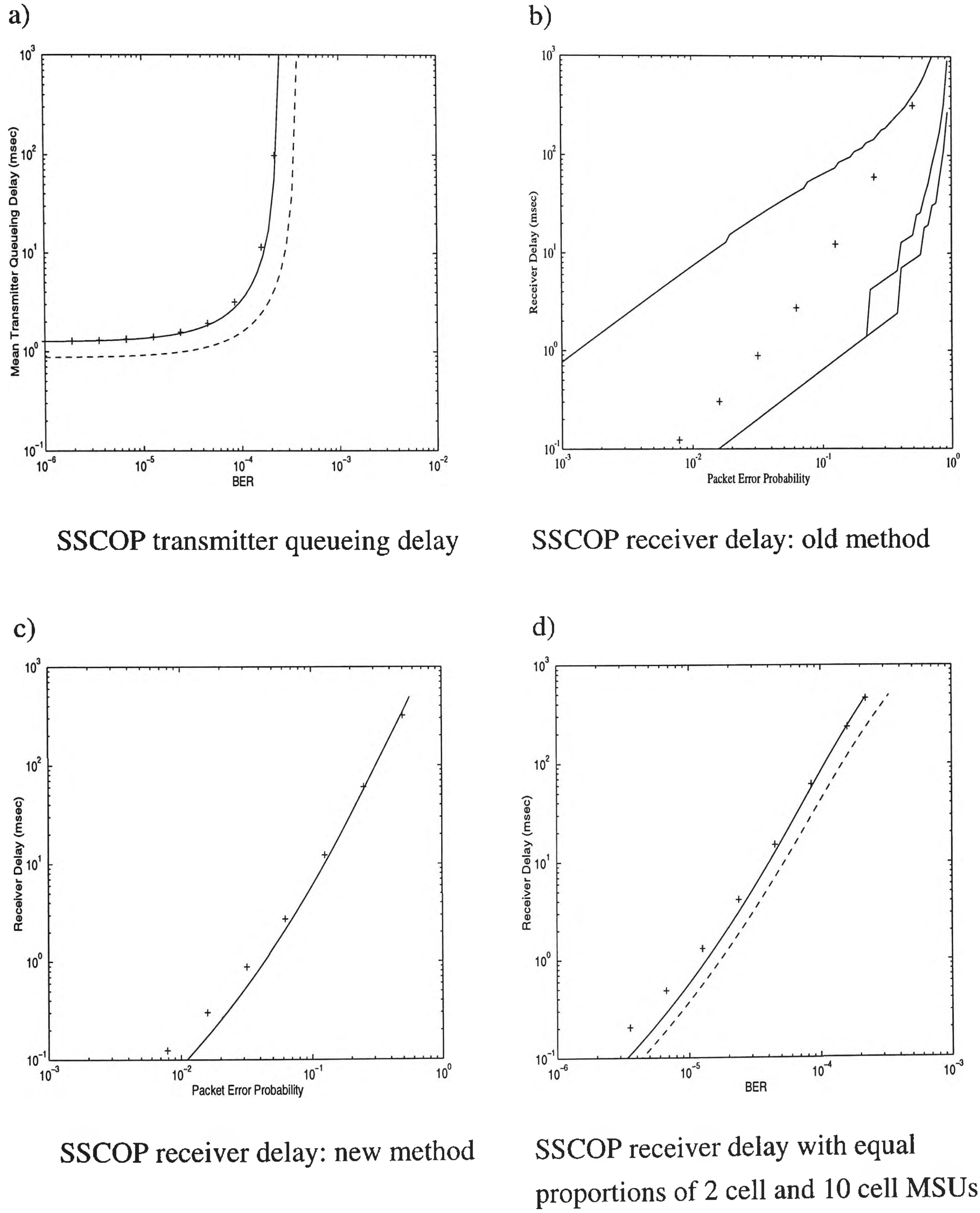


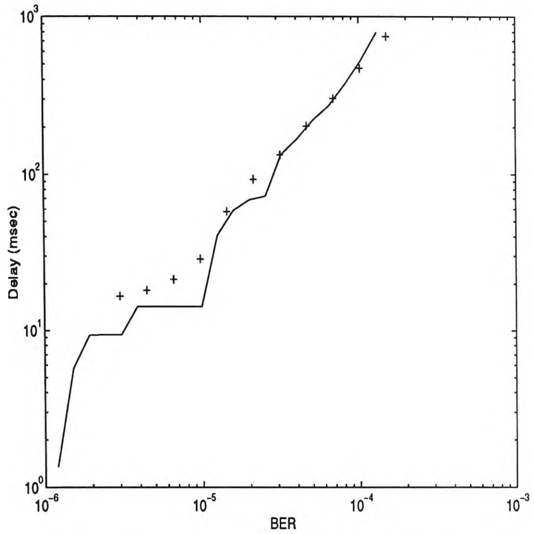
Figure 4.2 SSCOP mean delay results. Solid line indicates analytical results. Dotted line shows analytical results assuming fixed MSU sizes (i.e. the mean). "+" indicates simulation results.

Although Figure 4.2c (the new SSCOP receiver delay analysis) shows a good match between analytical and simulation results when the receiver delay becomes appreciable (i.e. more than 10 msec), there is some divergence when the delay is lower than this. For the purpose of determining error monitoring parameters (which has motivated the SSCOP performance modelling) these inaccuracies at low delays are not significant. As indicated in the previous section, they are due the modelling assumption of Poisson arrivals at the receiver. We therefore conclude that this assumption does not appreciably affect mean receiver delays in the region of interest (i.e. more than 10 msec).

Figure 4.2d shows the effect of different message size distributions on receiver delay. The upper plot (and the simulation points) show the receiver delay for the previously described mix of MSUs (i.e. 2 and 10 cell). The lower plot keeps all MSUs at the mean value (6 cells). As shown by the figure, this causes the mean receiver delay to be underestimated. The increase in receiver delay seen when the difference in message size is modelled (i.e. the upper plot) is due to the increased error probability experienced by the long (i.e. 10 cell) MSUs. Once again the divergence at low delay values is due to the assumption of poisson arrivals at the receiver.

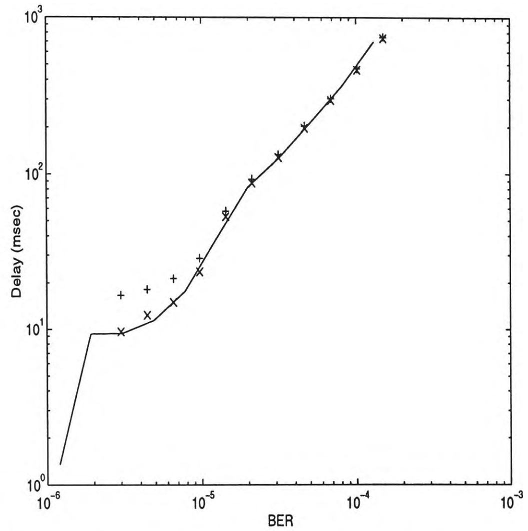
Figure 4.3a shows the 99th percentiles of the overall SSCOP delay, using the approximate distribution $F_s(t)$ given in Equation 4.22 (recall that this does not incorporate the queueing delay or the initial service (or transmission) time). While these results match the simulated SSCOP delay percentiles reasonably well at delay values exceeding 100 msec, there are appreciable inaccuracies at lower delays.

Figure 4.3b incorporates the refinements in section 4.4.2, resulting in a much closer match with the simulated results. Figure 4.3b also shows, using the 'x' signs, the simulated percentiles of the delay which is actually modelled by the improved version of Equation 4.22 (i.e. from when the initial MSU transmission completes until the packet is released from the receiver buffer, a)



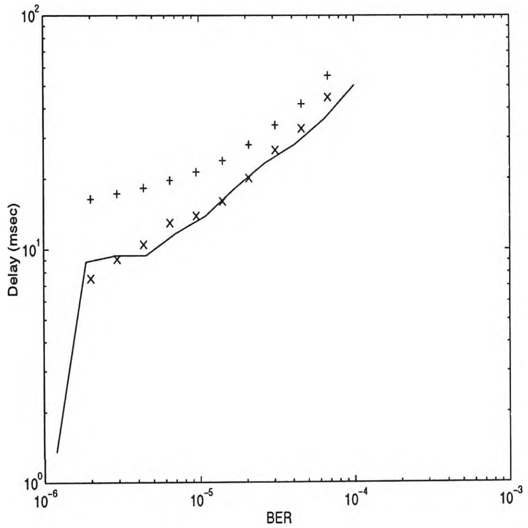
99th percentile using approximate method. Poll interval = 100 msec.

b)



99th percentile using improved method. Poll interval = 100 msec.

c)



99th percentile using improved method. Poll interval = 10 msec.

Figure 4.3 SSCOP delay percentile results. Solid line shows analytical results. "+"shows simulated results for the total delay. "x" shows simulated results for the portion of the delay captured by the model.

minus the propagation delay). As previously, the simulated percentiles of the total delay are shown by the '+' signs.

Figure 4.3c shows the 99th delay percentile results when the poll interval is reduced from 100 msec to 10 msec. Once again the '+' signs show the simulated 99th percentile of the total SSCOP delay, while the ‘x’ signs show that portion which is captured by the model (the improved version is used). In this case, where the queueing delay is a greater component of the overall delay, the model underestimates the actual SSCOP delay percentile. It is worth noting however that the poll interval for ATM signalling links is likely to be 100 msec [ITU93.2], where the analytical results are a good match to the simulation results.

4.6 Conclusions

This chapter has examined ATM signalling VCC delay performance at high error rates, motivated by the need to determine maximum error rates for given delay constraints. These maximum error rates form the basis of the ATM signalling error monitor design presented in chapter 5.

The delay performance investigation has focused on the Service Specific Connection Oriented Protocol (SSCOP), the link level protocol for the Signalling AAL. This uses a Selective Repeat ARQ. Chapter 3 revealed two simplifying assumptions used in previous Selective Repeat analyses; that round trip delays remain constant for packets needing multiple retransmission attempts, and that all packets have the same size (and hence the same error probability). This chapter examined the effect of these assumptions on SSCOP delay performance modelling. Our results have shown that:

- due to the difference (in the SSCOP) between the time to report an initial error and the time to report a failed retransmission, assuming constant round trip times leads to significant inaccuracies in receiver delay calculation. The same conclusion, although not tested here, applies (for the same reasons) to the calculation of the virtual service time (i.e. the time for a successful transmission to complete).

- assuming constant packet sizes (and hence constant error probabilities) causes mean queueing delay at the transmitter, and mean receiver delay, to be underestimated.

The new SSCOP delay model, by relaxing these two assumptions, overcomes these deficiencies. In addition, two other refinements are included which account for:

- the delay between the arrival of an errored packet at the receiver and its subsequent detection.

- the time for the server to become clear before a retransmission is sent (retransmitted packets have non-preemptive priority over other packets).

As ITU recommendation E.733 gives specifications for the 99th delay percentile (in addition to those for mean delay), two procedures (one approximate, the other more exact) for determining these percentiles have been presented. These procedures ignore the contribution of the transmitter queueing delay, to maintain tractability. While this causes the delay percentiles to be underestimated when the poll interval is small (e.g. 10 msec), accurate results (particularly with the more exact procedure) are obtained for the poll interval suggested for ATM signalling links (100 msec).

Recommendation E.733 also gives specifications for the sensitivity, with respect to utilisation, of the mean delay and the 99th delay percentile. As the delay expressions developed here cannot be expressed as a function of the utilisation (unlike the ITU SS7 delay formula), closed form expressions for these sensitivities are not practical. We therefore generate the required sensitivities numerically.

The SSCOP delay expressions developed in this chapter, by overcoming the deficiencies of previous methods, are well suited for their intended task, i.e. determining maximum sustainable error rates for ATM signalling links.

These maximum error rates, and the ensuing design of an ATM signalling error monitor, are the topics of the next chapter.

5. ATM Signalling Error Monitoring

5.1 Introduction

Chapter 4 introduced new methods for determining the delay performance of ATM Signalling Links. These methods give the maximum error rates which these links may sustain while satisfying given delay criteria. An ATM Signalling Error Monitor, which detects when this maximum error rate has been exceeded, is the topic of this chapter.

Section 5.2 outlines the requirements for ATM Error Monitors. These requirements form the basis of the comparisons which follow. Section 5.3 uses the modelling techniques of the previous chapter to establish maximum sustainable error rates for ATM signalling links, based on delay requirements. Section 5.4 looks at the options for error rate measurement, given the current structure of the Signalling AAL and the ATM Layer.

Sections 5.2 to 5.4 build upon the work of previous chapters to provide a framework for comparing ATM error monitors, in particular the 3 schemes listed at the end of chapter 3 (i.e. the Errored Interval Monitor (EIM), the Leaky Bucket and the OAM monitor). Section 5.5 examines each of the schemes in more detail, emphasizing methods for finding parameter values. In particular, a proposal is outlined for an OAM based ATM Signalling Error

Monitor. Section 5.6 shows how to determine the mean changeover time for a given error rate, for both uniform (i.e. independent) and bursty errors.

The results in section 5.7 show that while the proposed OAM scheme gives the best performance of the three error monitors, it does not detect bursty error conditions. An enhancement to the proposed OAM monitor thus is outlined, which allows it detect to detect bursty errors.

5.2 Performance Monitoring Requirements

Chapter 3 presented three requirements for "safe and effective" SS7 error monitors (as proposed in [SCH94]). Then 3 potential ATM error monitors were presented. These error monitors are compared in this chapter, on the basis of how well they meet these 3 requirements, i.e. that they

1) cause a link changeover when the maximum error rate is exceeded.

2) tolerate burst errors for a specified period of time before initiating changeover.

3) ensure that the "changeover transient" (i.e. the amount of data queued when changeover is initiated) does not exceed a specified value.

In the ideal case, an error monitoring scheme should meet these criteria, irrespective of the link operating points (i.e. peak rate, MSU length, utilisation, propagation delay). Accordingly we shall examine the performance of ATM error monitors over a range of link operating points, which are as follows:

Peak rate

ATM signalling peak rates are likely to be greater than their SS7 counterparts, due to the larger signalling messages in ATM (for the more complex call types planned in Capability Set two, an increase in message size of about a factor of 4 is likely [ITU93]). Hence a similar increase in the signalling peak rate is also likely. We consider 2 peak rates, 1 Mbps and 5 Mbps. The 1

Mbps example represents the ATM equivalent of a SS7 link set with four 64 kbps links (with the peak rate scaled up to accommodate the longer ATM signalling messages), while the 5 Mbps example represents the ATM equivalent a single high speed (1.536 Mbps) SS7 link.

MSU Length

While future B-ISDN signalling patterns are unknown, and will depend on the requirements of new services (e.g. database accesses, User to User information), it is likely that the range of messages sizes will be greater than for SS7 (which carries mostly ISUP messages). To investigate the effect of different traffic distributions, we shall consider the following traffic mixes.

1) Equal proportions of 10 cell and 2 cell MSUs. This corresponds to the approximate size of query and response messages for Capability Set 2 signalling messages [ITU93].

2) All 2 cell MSUs. These would come from an SP responding to a sustained burst of incoming calls, while generating none of its own.

3) All 20 cell MSUs.

Utilisation

A signalling link must be able to carry the load of a failed link, in addition to its own load [ITU92.2]. If we assume a maximum link utilisation of 0.4 before a changeover (as in [KAN93.2]), then in a worst case (i.e. after a changeover and at maximum load), a link will have a utilisation of 0.8. Hence we shall test the respective ATM error monitoring schemes at this utilisation.

As SS7 links transmit Fill In Signal Units (FISUs) continuously when there is no other traffic to send, they maintain a constant bit rate (however FISUs are not retransmitted in case of error). Hence the rate at which the SUERM counts errors, for a given BER, is not affected greatly by the offered load (with the EIM, the offered load does not affect this rate at all).

As mentioned in chapter 2, FISUs are not planned for ATM signalling links. This will effect error monitor performance, as the rate at which errors are counted, for a given BER (or cell error rate), will depend on the offered load. As this rate will determine the changeover time, we would therefore expect ATM error monitors to change over more quickly when the offered load increases (unlike SS7 error monitors). To investigate this variability in changeover time, we shall examine ATM error monitor performance at a "normal" utilisation (i.e. 0.1), in addition to the worst case (0.8).

Propagation Delay

[KAN93.2] considers the SS7 EIM performance over a range of link lengths (and hence propagation delays), and concludes that 3 sets of parameters are needed, depending on whether a short, medium or long link is being considered. To simplify this investigation, we consider a worst case (i.e. longest) link only, with a round trip time of 50 msec. A Poll interval of 100 msec is also assumed (as previously).

The operating points chosen for this study, while reasonable, are also somewhat arbitrary, as there are no current ATM signalling links to act as a guide. Hence the parameter values arising from this set of operating points would need to be reviewed (using the techniques outlined in this chapter) for a real set of ATM signaling links, which may have a different range of operating points.

5.3 Maximum Sustainable Error Rate

Signalling links are required to operate so that the MSU delay does not exceed a specified set of constraints (i.e. those set out in ITU recommendation E.733: mean delay, mean delay sensitivity with respect to utilisation, 99th percentile of delay, and 99th percentile sensitivity). If any of these constraints are not satisfied, the link must be changed.

Unfortunately it is difficult to monitor delay performance directly. Thus a performance monitoring scheme must use an indirect measure of performance, such as cell or MSU error rate. This gives a maximum sustainable error rate (cell or MSU), which is the maximum error rate which a link may suffer while still satisfying delay constraints. Maximum sustainable error rates were discussed in chapter 3 (section 3.2.3) in the context of SS7 links. Here we determine maximum sustainable error rates for ATM signalling links, with the results being used in section 5.6 to determine parameter values for the respective ATM error monitors.

The current specifications for worst case SS7 delay performance are provided in ITU recommendation E.733 [ITU92.2] so that network designers may determine maximum SS7 link utilisations. In the absence of a similar standard for ATM signalling links, we shall use the SS7 figures here as well. E.733 specifies the maximum delay under worst case operating conditions, i.e. an error rate of 1/256 (the SUERM removes links from service at error rates higher than this). The aim is to dimension links so that at their maximum utilisation, and under these worst case error conditions, the delay figures are not exceeded. These figures (given previously in chapter 3) are: mean delay 60 msec, delay sensitivity (wrt utilisation) 300 msec/erlang, 99th delay percentile 300msec and 99th percentile sensitivity (wrt utilisation) 1500 msec/erlang.

Here we apply the above process in reverse, as our aim is to determine the maximum error rate for ATM signalling links (as opposed to 64 kbps SS7 links, where the maximum error rate is set by the SUERM parameters). Hence we specify a link (i.e. ATM signalling VCC) utilisation, a peak rate and a message size distribution, then determine the maximum cell (or MSU) error rate which still satisfies the delay constraints of E.733. We obtain these maximum error rates using the SSCOP delay performance modelling techniques of chapter 4.

Table 5.1 gives the maximum sustainable error rates for a traffic mix comprising 50% 2 cell MSUs and 50% 10 cell MSUs. The remaining link parameters are those given in the previous section. Similarly table 5.2 and table 5.3 give maximum sustainable error rates for the 2 cell MSU and 20 cell MSU cases respectively.

Peak (Mbps)	**Utilisa-tion**	**Max Cell error prob.**	**Max Msu error prob.**	**Con-straint**
1	0.1	2.62e-02	1.42e-01	99th
1	0.8	1.04e-02	6.01e-02	Mean del.
5	0.1	1.52e-02	8.63e-02	Del. sens.
5	0.8	6.19e-03	3.62e-02	Mean del.

Table 5.1 Maximum Sustainable Error Rates: 50% 2 cell MSUs, 50% 10 cell MSUs

As these tables show, the maximum sustainable MSU error rate (or error probability) for these ATM signalling links will be much higher than the corresponding error rate[1] for SS7 links (i.e. 4.0e-03). This is due to the ability of the Selective Repeat ARQ to withstand higher error rates than go-back-n, as shown by the example in chapter 3 (i.e. Figure 3.2).

While ATM signalling links may operate at much higher MSU error rates than their SS7 counterparts, the actual link requirements (in terms of BER) may actually be more stringent than for SS7, due to the increased MSU length expected for ATM. For example, the worst case BER (i.e. the highest before changeover) for a 64 kbps SS7 link is 1.63e-05, assuming a mean Sig-

1. While we shall generally refer to "error rates", the actual figures given will be error probabilities (i.e. 1/error rate).

nal Unit length of 30 octets. However the worst case BER for the ATM links shown here is 4.34e-07 (corresponding to a cell error probability of

Peak (Mbps)	**Utilisa-tion**	**Max Cell error prob.**	**Max MSU error prob.**	**Con-straint**
1	0.1	7.07-02	1.36e-01	Del. sens.
1	0.8	2.77e-02	5.47e-02	Mean del.
5	0.1	3.49e-02	6.85e-02	Del. sens.
5	0.8	1.25e-02	2.47e-02	Mean del.

Table 5.2 Maximum Sustainable Error Rates: All 2 cell MSUs

Peak (Mbps)	**Utilisa tion**	**Max Cell error prob.**	**Max MSU error prob.**	**Con-straint**
1	0.1	1.01e-02	1.83e-01	99th
1	0.8	3.72e-03	7.18e-02	Mean del.
5	0.1	1.04e-02	1.88e-01	99th
5	0.8	3.72e-03	7.17e-02	Mean del

Table 5.3 Maximum Sustainable Error Rates: All 20 cell MSUs

3.72e-03 for the 5 Mbps link carrying 20 cell MSUs).

One of the major results from [HOU94] is the insensitivity of the maximum sustainable error probability to changes in MSU size, for high speed links (i.e. 1.536 Mbps). The same however is not true for the ATM signalling links considered here, as shown in tables 5.2 and 5.3. Here the maximum sustainable cell error rate varies by more than an order of magnitude as the MSU size increases from 2 cells to 20 cells. This is due to two characteristics of the Selective Repeat ARQ: the probability of failed retransmissions (with a resulting increase in receiver delay) increases with the message size, in addition to the expected increase in transmitter delay.

Maximum sustainable error rates, such as those shown here, provide the criteria for link changeover. For a given peak rate, this changeover threshold is based on the most stringent sustainable error rate (i.e. the worst case). However, as shown in Table 5.2 and Table 5.3, this worst case will depend on whether MSU error rates or cell error rates are monitored. Considering the 5 Mbps example from these tables, the changeover MSU error probability (i.e. the most stringent result from the three tables) is 2.47e-02, the results for 2 cell MSUs at a utilisation of 0.8. In contrast to this however, the changeover cell error probability (3.72e-03) comes the results for 20 cell MSUs (at a utilisation of 0.8). A key observation is that if MSU error rates are the basis for performance monitoring, then the full range of MSU sizes (rather than just the largest) must be considered to determine the maximum sustainable error rate.

We shall consider using both MSU error rates and cell error rates to determine when a link changeover is needed. The next section examines the protocol mechanisms which enable such error rates to be measured.

5.4 Measuring Error Rates

Here we examine the options for detecting and reporting errors (both cell and MSU) in an ATM environment. We begin by reviewing how errors are detected in SS7.

5.4.1 SS7

The SS7 SUERM identifies two error conditions: loss of individual Signal Units, indicated by a CRC error or an incorrect length (i.e. too long or too short), and loss of alignment. During this latter condition, which implies a complete failure of the transmission system, the SUERM enters the "octet counting" mode, where the error counter is incremented for every 16 octets that pass. A key feature with SS7 error detection (in the context of this discussion) is that the required measurements (counting of Signal Units, errored

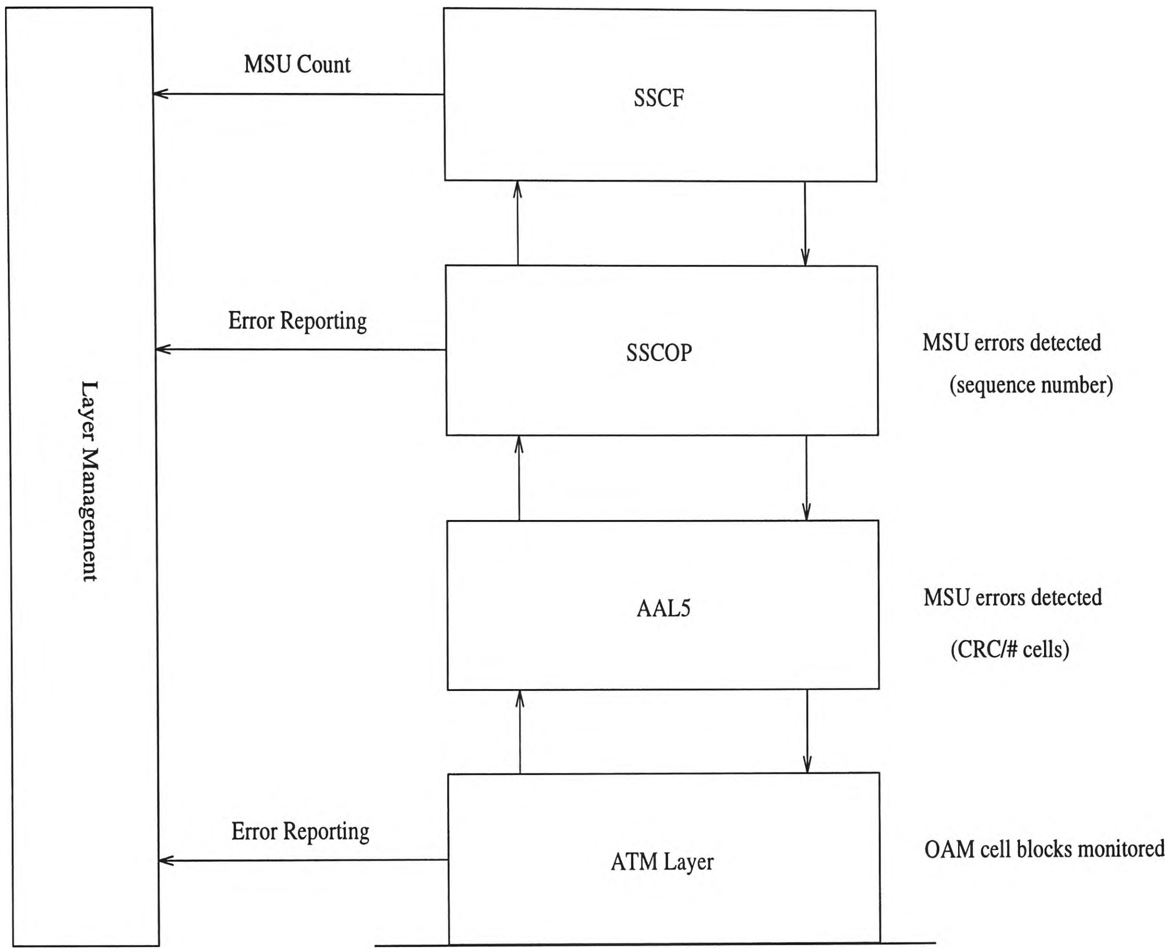


Figure 5.1 Error Detection and Reporting

or otherwise and checking for loss of alignment) all occur at the same place, i.e. the boundary between MTP1 and MTP2.

5.4.2 ATM Cell Errors

Here we review the error detection options available for ATM, concentrating first on cell error detection.

While all SS7 error conditions are detected at the same place, the error detection mechanisms for ATM signalling occur in various (and separate) parts of the protocol stack. We begin by considering cell errors. Figure 5.1 shows the Signalling AAL and the ATM Layer, indicating where errors of various types are detected. The ATM layer checks for errors in the cell header, and provides single bit error correction. However if two errored cell headers are detected back to back, the second cell is discarded. Any subsequent cells with errored headers are discarded as well, until a correct cell header is detected. Hence during an error burst (where we may assume that every cell header is corrupted), the errored cells are discarded at the ATM Layer, and do not arrive at the Signalling AAL.

To monitor error performance at the cell level, the OAM procedures described in ITU recommendation I.610 [ITU93.4] are used, as indicated in chapter 2. These operate by inserting a special OAM cell after each block of cells in a VCC, with a minimum recommended block size of 128 cells. This OAM performance monitoring cell indicates the number of cells in the block (to allow cell loss detection) as well as providing a parity check over the entire block. Sequence numbers are used to detect lost OAM cells. If the link becomes idle, OAM continuity cells are sent at regular intervals (T_c). If no continuity (or user) cells are received for an interval exceeding $2T_c$, then the Virtual Channel Connection is presumed to have failed

As the OAM scheme operates on cell blocks, rather than individual cells, it cannot accurately measure the cell error rate. While lost cells may be counted (by comparing the number of cell arrivals to the block size), the number of cells per block lost due to bit errors cannot be determined exactly (this would require each cell to be tested). To avoid any ambiguity in the number of errored cells, we propose instead to record cell blocks as either errored or error free. The resulting error monitoring algorithm (which we describe in the next section) is similar to that used for MSUs, with each cell block being equivalent to a fixed size "MSU".

OAM cells carry a sequence number (modulo 8), which may be used to detect any which are lost. With our proposed error monitoring scheme, any missing OAM cells (detected by a skip in sequence numbers) would be recorded as errored blocks. The results from each OAM cell block is relayed to the transmitter by OAM cells travelling in the reverse direction. This allows the error monitor to be located at the transmitter, if desired. However, to avoid this extra delay in reporting results (as well as to avoid the possibility of the reverse OAM cells being lost), we propose that the OAM based error monitor be located at the receiver.

5.4.3 ATM MSU Errors

While the ATM layer considers cell level errors, the Signalling AAL detects Protocol Data Unit (PDU) errors (i.e. MSU errors). This is done in two places, as shown in Figure 5.1. The AAL5 checks PDU[2] integrity using the CRC and the PDU length indicator. Errored PDUs are discarded (the AAL5 standard has an option which allows errored PDUs to be passed to the higher layer, with an appropriate indication. This will not be used for signalling). The standard implementation of AAL5 has no procedure for reporting PDU arrivals (or lost PDUs) to layer management [ITU93.4]; such a procedure would be needed if signalling error monitoring were to be done at this level. In addition, AAL5 does not measure the length of error bursts, as it has no knowledge of missing PDUs. Hence AAL5 would interpret an error burst as consisting of two errors only, the PDUs at the start and the end of the burst (in a worst case, where the cells lost in a burst remove an exact (i.e. integral) number of PDUs, the AAL5 would detect no errors at all).

All PDUs which are accepted by AAL5 are passed to the SSCOP, which will detect errors in the initial transmissions, due to a skip in sequence numbers. Hence the SSCOP (unlike AAL5) can measure how many errors there are in a burst by examining the gap in the sequence numbers. However the SSCOP receiver does not detect errored retransmissions. These are detected instead at the transmitter by way of the poll procedure, as described in chapter 4 (section 4.2). In a worst case scenario, when only retransmissions were being sent, the SSCOP receiver would record no errors at all. The SSCOP also has a procedure for detecting a total link failure, which is declared if no POLL messages are sent for an interval set by the Timer_NORESPONSE. Given this "keep-alive" feature, the continuity check provided by the OAM mecha-

2. As AAL5 and the SSCOP are generic (i.e. not specific to signalling) protocols, we shall refer to PDUs (rather than MSUs) during this section.

nism should either not be used, or the timeout interval be set to greater than Timer_NORESPONSE.

It is clear that neither the AAL5 nor the SSCOP are able to detect all errored PDUs. However this may be done at the SSCOP transmitter, which knows how many retransmissions it has done (with Selective Repeat, each PDU error causes one retransmission (unlike go-back-n)). It is not surprising then that [ITU94.4] has proposed that MSU error rate monitoring be done at the transmitter. To determine an error rate using this method, the total number of transmissions is also needed. The method proposed in [ITU94.4] is for the Service Specific Coordination Function (SSCF) to indicate to the layer management how many MSUs it has passed to the SSCOP (as shown in Figure 5.1). This number, combined with the error count, provides the required MSU error ratio.

We foresee a potential problem with counting errors at the SSCOP transmitter when an EIM monitor is used. This is because the temporal relation between successful transmissions and errors is lost (essentially the transmitter records errors as they are reported in acknowledgements from the receiver, and not as they actually occur). This will affect the EIM, which needs to know exactly when errors occur, so that an interval may be recorded as errored or otherwise. In a worst case situation, when only retransmissions are being sent, errors are reported only when poll messages are sent (e.g. every 100 msec). If, for example, the EIM interval was 20 msec, then this error reporting scheme would severely under-estimate the interval error rate, as only 1 interval in 5 would be errored. This problem may be avoided however if the poll interval is made the same as the EIM interval.

The purpose of this section, and the previous section, has been to provide the groundwork to allow a comparison of the performance of the three error monitoring schemes presented in chapter 3. We now consider how each of these schemes will work in an ATM environment.

5.5 Error Monitor Operation in ATM

Here we consider each of the 3 error monitoring schemes outlined in chapter 3, and show how to determine parameter values which satisfy the error monitor requirements of section 5.2. We begin with the EIM.

5.5.1 EIM

The basic operation of the EIM was outlined in chapter 3. We now describe how it may be adapted to ATM signalling links, and how parameter values should be chosen.

Due to the transmission of Fill In Signal Units, each EIM interval in SS7 has the same number of bits. However the variation in the number of bits (i.e. cells) per EIM interval in ATM will require a modification to the EIM implementation, to account for intervals which have no cells at all. When such an interval occurs, the EIM counter should remain unchanged, as there is no new information regarding link performance.

As previously indicated, the most appropriate location for error monitoring at the SSCOP level is at the transmitter. Hence an ATM EIM implementation would be located at the transmitter, as part of the Layer Management for the Signalling AAL. The EIM would count an errored interval as one during which a retransmission indication arrives from the SSCOP (in the SSCOP specification, this is a MAA-ERROR.indication signal with an error code of V). An interval with no cells is one during which the Layer Management receives: a) no MAA-ERROR.indication signals from the SSCOP and b) no transmit indications from the SSCF (which are sent using the MAA-DATA-COUNT.indication signal).

For the EIM to count all errored intervals at high error rates, the poll interval would need to equal the EIM interval, as indicated in section 5.4. The need to match errors to their respective time intervals also places a lower limit on the EIM interval. Essentially this interval must be large enough to allow for any delay jitter between the time that a retransmission is detected by the SSCOP and the sending of the appropriate MAA-ERROR indication (particularly if a burst of retransmissions causes a number of these MAA-ERROR indications to be queued). The size of this delay jitter is difficult to quantify, as it depends on the SSCOP implementation. For the purpose of this investigation, we assume (arbitrarily) that the smallest practical EIM interval is 20 msec.

Methods for determining parameter values for the SS7 EIM (i.e. U and D, the respective addition and reduction to the EIM counter for errored/error free intervals and T, the counter value at which changeover occurs) have been given in chapter 3 (section 3.4.3). We now determine the corresponding parameter values for an ATM EIM.

The first requirement for the EIM (and other ATM error monitors) is to changeover as soon as possible after the maximum sustainable cell error rate (p_{cmax}) is exceeded. A given value of p_{cmax} will set a maximum sustainable interval error probability, p_{imax}. To obtain p_{imax} we make a simplifying assumption, namely that the number of cells in each interval has a constant value N (the mean)[3]. Then p_{imax} is just $1 - (1 - p_{cmax})^N$, assuming independent cell errors. Note that at low traffic levels, when N is small, the value for p_{imax} will decrease (i.e. become more stringent).

To ensure a changeover when the interval error probability exceeds p_{imax}, we choose U and D so that $D/(U+D) = p_{imax}$ (note that U and D must be scaled to give integer values). When determining p_{imax} however, we observe a difference between the SS7 and the ATM EIM implementations. As the number of bits per interval is constant for the SS7 EIM, the interval error probability is determined by the error rate only. However for ATM links, the number of bits (or cells) per interval varies with the utilisation, so that the interval error probability is a function of the error rate and the utilisation.

3. To test this assumption, we calculated p_{imax} for the scenario where the number of cell arrivals during an EIIM interval had a poisson distribution with the same mean. The p_{imax} values obtained using the poisson arrival assumption were almost identical to those obtained when the number of arrivals is fixed.

The values for U and D, which enforce p_{imax}, must ensure that changeover occurs at the correct p_{cmax} value at all utilisations (noting that there will be a different value for N, p_{cmax}, and hence p_{imax} for each utilisation. U and D must be chosen to satisfy the most stringent p_{imax} value over the range of utilisations considered). This in turn ensures that the cell error rate at which changeover occurs allows the delay specifications to be met at all utilisations. Note that for SS7 links, values of U and D which satisfy delay specifications at the maximum utilisation will apply to lower utilisations as well.

The requirement to ride over occasional short error bursts, where every interval is errored (i.e. requirement number two), will be satisfied for the ATM EIM by an appropriate choice of interval size τ, U and T (i.e. $\tau(T/U) > maximum\ burst\ length$). In effect this requirement places a lower bound on T, for given values of U and τ.

Similarly the need to limit the changeover transient (the third requirement) places an upper bound on T. During a worst case error situation, when the link is carrying retransmissions only (i.e throughput drops to zero), the transient will be the amount of data which arrives while the EIM counter moves from 0 to T. Hence the maximum mean changeover transient is the product of the time t_{co} to reach T under worst case conditions (i.e. every interval errored) and λ_{max}, the maximum arrival rate (i.e. at the highest utilisation) in bits/sec. Accordingly, the upper bound for T is the highest value which satisfies the relation $\tau(T/U) < t_{co}$ where $t_{co}\lambda_{max}$ is the maximum mean changeover transient allowed. For the examples in the following section, we shall set T to this upper bound.

The remaining parameter to be chosen is the EIM interval, τ. [SCH94] chooses a value of 100 msec for 1.536 Mbps. We use this figure as a guide for our investigation, and hence choose (initially) an EIM interval of 100 msec for the 1 Mbps link, and scale this to 20 msec for the 5 Mbps link. The suitability of this value will be investigated further in section 5.7.

We now use the above techniques to determine parameter values for the 5 Mbps and 1 Mbps links described previously. For the purpose of this example, we set maximum mean changeover transients of 2 Mbit and 0.4 Mbit respectively. This corresponds to a maximum changeover time (under worst case error conditions) of 500 msec, assuming a maximum utilisation of 0.8. We assume that delay specifications are to be satisfied for utilisations ranging from of 0.1 to 0.8 (i.e. the values given in the tables). Using the procedures outlined above, the resulting parameter values are U = 18, D = 5 and T = 450 for the 5 Mbps link, and U = 11, D = 3 and T = 55 for the 1 Mbps link.

5.5.2 MSU Based Monitor (SUERM)

Here we outline the operation of a performance monitor which counts MSUs. This monitor is equivalent to the SS7 SUERM.

The SS7 SUERM uses a Leaky Bucket to monitor link performance. This increments a counter by 1 for every error, and decrements the counter by 1 for every D_{lb} MSUs which pass (actually every D_{lb} Signal Units which pass, as Fill In Signal Units (FISUs) are counted as well). D_{lb} is chosen so that $1/D_{lb}$ equals the changeover error threshold. The Leaky Bucket signals a link changeover when the counter reaches a level T_{lb} (for SS7 links, T_{lb} is 64 and D_{lb} is 256).

We now consider parameter values for an SUERM which runs on ATM links. To find U, for a given peak rate, we must consider the worst case MSU error probability (over all utilisations and MSU sizes). This becomes the changeover error threshold. Referring to tables 5.1, 5.2 and 5.3, we have a worst case MSU error probability (p_{mmax}) of 2.47e-02 and 5.47e-02 for the 5 Mbps and 1 Mbps links respectively. This is for the 2 cell MSUs with a utilisation of 0.8. This gives D = 41 for the 5 Mbps link and D = 19 for the 1 Mbps link.

When determining T, we encounter two conflicting requirements. T must be a) large enough so that a short error burst does not cause a changeover and b) small enough so that the changeover transient is below a pre-determined limit. We begin by limiting the changeover transient. As with the EIM, we consider a worst case error situation, where the link sends retransmissions continuously (i.e. throughput drops to 0). Similar to the EIM example, we specify maximum changeover transients of 2 Mbit and 0.4 Mbit for the 5 Mbps and 1 Mbps links respectively (i.e. a changeover after 500 msec during this worst case condition). This assumes that the offered load is 0.8 of the link capacity (i.e. the utilisation is 0.8), and that all of it is queued. At a link rate of 5 Mbps, the number of MSUs sent during 500 msec is 294.8, if we assume 20 cell MSUs (i.e. a worst case). This assumes that the MSUs are transmitted continuously on the link. We also assume that in the worst case, each of these 294.8 MSUs is errored. Hence we choose T to be 294. Using a similar argument, we get T = 59 for the 1 Mbps link.

However these values of T will not satisfy the minimum changeover time requirement, due to variations in MSU length. For example we may require a minimum changeover time of 400 msec during an error burst (as for the EIM example). If we now assume 2 cell MSUs (the worst case in this situation), and assume again that the total link utilisation goes to 1 (due to retransmissions), then T should be at least 2358 for the 5 Mbps link (i.e. the number of two cell MSUs sent in 400 msec).

Clearly it is not possible to satisfy both requirements (i.e. minimum changeover time, limiting the changeover transient) with a single value for T, due to the change in Leaky Bucket performance for different MSU sizes. To overcome this problem, a timer could be coupled to the Leaky Bucket. This timer would start whenever the Leaky Bucket counter started counting up from 0, with a changeover not being permitted until the timer had reached a minimum value (e.g. 400 msec). The timer would be reset when the Leaky Bucket counter returned to 0. We note however that this problem would not arise if all MSUs were the same size. This is essentially the thrust of the OAM based scheme, which we consider next.

5.5.3 Proposed OAM Based Monitor

The OAM based error monitor which we propose uses a similar counter to the EIM. However, rather than considering time intervals, the counter increases instead by U for each errored block and decreases by D for each error free block, with a changeover at a level T.

To determine U and D, we consider the maximum sustainable block error rate, $p_{bmax} = 1 - (1 - p_{cmax})^N$, where N is the number of cells in the block. Following the approach of the EIM, we choose U and D so that $D/(U+D) = p_{bmax}$. To allow comparison between the OAM results and the EIM results, we choose (initially) a block size equal to the mean number of cells in the EIM intervals chosen earlier, i.e. 20 and 100 msec (for the 1 Mbps and 5 Mbps links respectively). This gives an OAM block size of 189 cells, for both the 5 Mbps and the 1 Mbps links. Using the p_{cmax} value of 3.72e-03 from Table 5.3 (for 20 cell MSUs and a utilisation of 0.8), we get U = D = 1 for both the 1 Mbps and 5 Mbps links.

As all OAM blocks on a given link have the same size, a single value of T satisfies both the maximum changeover transient and the minimum changeover time requirements (unlike the MSU based scheme). This assumes that the time to reach the maximum changeover transient under worst case error conditions exceeds the minimum changeover time. Once again we assume that a changeover must occur within 500 msec during a worst case error situation, (i.e. 100% error probability). The number of 189 cell blocks sent on the 5 Mbps and 1 Mbps links in 500 msec, assuming continuous transmissions, is 6 and 31 respectively. As U = 1 for this example, T is 6 for the 1 Mbps link and 31 for the 5 Mbps link.

The OAM error monitor which we propose requires all cell blocks to be the same size. However this is not guaranteed by I.610, which says that “a performance monitoring cell insertion request is initiated after every N user cells. The monitoring cell is inserted at the first free cell location after the request ... the monitoring cell must be inserted into the user cell stream no more than N/2 user cells after an insertion request has been initiated". In particular, during a worst case error situation, when the link utilisation (including retransmissions) may be 1, there may not be a free cell location after the Nth cell. However it is during this worst case scenario that the requirement for a constant OAM cell block size is most pressing, in order to limit the changeover transient. During the discussion (and the resulting proposals) which follow, we shall assume that the OAM implementation does allow the forced insertion (if necessary) of an OAM cell after every N user cells, thereby guaranteeing a constant OAM cell block size.

This section has outlined methods for choosing parameter values for each of the three error monitoring schemes. These methods have been applied to the 1 Mbps and 5 Mbps links considered in our study. However to compare the respective error monitor performances, we need methods to find the mean changeover time, as a function of the error rate and the link parameters. These methods are the topic of the next section.

5.6 Changeover time

A method for determining the mean time to overflow a Leaky Bucket (or SUERM), as a function of the MSU error rate, has been presented in chapter 3 (section 3.4.2) and appendix 1 (note that to retain tractability, the SUERM analysis assumes constant MSU sizes, unlike the delay performance analysis of the previous chapter). However, to compare the performance of the three ATM error monitors considered in this chapter, we also need to find the changeover time for the EIM and the OAM monitors. Methods for doing this are presented here.

To model the performance of the EIM and OAM schemes, we adapt the method of [KAN93.1]. This method, which was developed for the SS7 EIM, uses an absorbing Markov Chain, similar to our Leaky Bucket analysis of chapter 3. The approach for the EIM (which we consider first) is to consider a Markov Chain with T+1 states, where each state corresponds to a level of the EIM counter (including 0). Decisions regarding change of state are made after each EIM interval. For each state x (except T, the absorbing state) there are three possibilities, which are

1) The state (i.e EIM counter level) remains the same. This event, which corresponds to an EIM interval where no cells arrive, occurs with probability $p_{same} = e^{-\lambda_c \tau}$, where λ_c is the mean cell arrival rate and τ the interval length. We assume that the cell arrivals are poisson.

2) The state goes to $x + U$, or T if $x + U > T$. This event, which corresponds to an errored interval, occurs with probability $p_{up} = (1 - p_{same})\left(1 - (1 - p_c)^N\right)$. Here p_c is the cell error probability and N the mean number of cells per interval.

3) The state goes to $x - D$, or 0 if $x - D < 0$. This event, which corresponds to an interval with cells (but no errors), occurs with probability $p_{down} = (1 - p_{same})(1 - p_c)^N$.

Using these probabilities (i.e. $p_{same}, p_{up}, p_{down}$), we form a transition matrix P. Similar to the approach in chapter 3, we form a sub-matrix Q by removing the T+1th row and column of P, and define N_{co} as the inverse of $I - Q$, where I is the identity matrix. The mean number of steps (i.e. EIM intervals) to overflow the EIM counter is the sum of the elements of the first row of N_{co}. The mean EIM changeover time is the product of the number of steps and τ, the interval length.

As we are interested in the changeover performance at very high error rates, we need to account for the extra traffic due to retransmissions. To do this, we find the MSU error probability, i.e. $p_m = 1 - (1 - p_c)^l$, where l is the mean MSU size (in cells). The total rate at which MSUs are sent (including retransmissions) is $\lambda_{tot} = \lambda(1 + p_m/(1 - p_m))$, where λ is the arrival rate of new MSUs. This gives a total cell transmission rate of $\lambda_{cell} = \lambda_{tot} l$. The mean number of cells per EIM interval is then $N = \lambda_{cell} \tau$

The changeover time analysis for the OAM error monitor is similar to the EIM case. Once again we construct a Markov Chain with T+1 states, however in this case there are only two possible moves from each state x, i.e.

1) The state goes to $x + U$, or T if $x + U > T$. This event, which corresponds to an errored block occurs with probability $p_{up} = 1 - (1 - p_c)^{N_{OAM}}$. Here N_{OAM} is the OAM block size (in cells).

2) The state goes to $x - D$, or 0 if $x - D < 0$. This event, which corresponds to an error free block, occurs with probability $p_{down} = (1 - p_c)^{N_{OAM}}$

Using the same method as before, we obtain the mean number of OAM blocks until changeover, which we denote as n_{blocks}. The mean changeover time is then $(n_{blocks} N_{OAM}) / \lambda_{cell}$

The mean changeover time analysis for the EIM and OAM schemes is easily extended to the bursty error case. While [KAN93.1] considers bursty errors briefly, extensions to the analysis to account for bursty errors are not presented. [RAM93] considers bursty errors in his SUERM model, but does not present the details of the analysis. Here we present the bursty error analysis, for the OAM case. The error model we use alternates between a high state (with a BER of b_h) and a low state (with a BER of b_l). To maintain tractability, we assume that the error state does not change during a block transmission time (a similar assumption is made in [RAM93]). At the end of each OAM block (with a high error state), the error state remains high for the next block with probability p_{high}. A similar probability, p_{low}, exists for the low error state.

For the results which follow (in section 5.7), we assume that the mean time in the high and the low error state is known. The task is then to find p_{high} and p_{low}. To do this we make a simplifying assumption, namely that the mean

OAM block transmission time, t_{OAM}, is less than the mean time in the high (or low state). We then get:

Mean time in the high state = $t_{OAM}/(1-p_{high})$ and mean time in the low state = $t_{OAM}/(1-p_{low})$.

The above assumption, used to obtain p_{high} and p_{low}, means that changeover times (for bursty errors) can only be obtained for a limited set of cases (i.e. when the mean OAM block time is less than the mean time in the low and the high error state). While it would be desirable to obtain an expression which gives the changeover time for all cases, the more limited expression outlined here is sufficient for the needs of this thesis, namely to highlight the potential shortcomings of error monitors under bursty error conditions.

We form a Markov Chain as previously, however each counter level (which before corresponded to a single state in the Markov Chain) is now associated with an error state as well. This effectively doubles the total number of states. If we define p_{chigh} as the cell error rate in the high error state (and similarly p_{clow} for the low error state), then there are 4 possible moves from a given state. If we consider the Markov Chain to be in the state x_{high} (i.e. counter level x and the high error state), then these 4 moves are

1) Move to the state $(x+U)_{high}$ (or T if $x+U>T$) with probability $p_{high}\left(1-(1-p_{chigh})^{N_{OAM}}\right)$.

2) Move to the state $(x+U)_{low}$ (or T if $x+U>T$) with probability $(1-p_{high})\left(1-(1-p_{chigh})^{N_{OAM}}\right)$.

3) Move to the state $(x-D)_{high}$ (or 0 if $x-D<0$), with probability $p_{high}(1-p_{chigh})^{N_{OAM}}$.

4) Move to the state $(x-D)_{low}$ (or 0 if $x-D<0$), with probability $(1-p_{high})(1-p_{chigh})^{N_{OAM}}$.

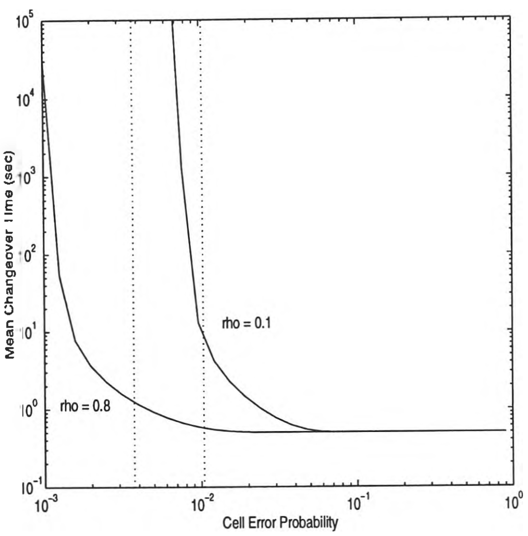
A similar set of probabilities exist for the Markov Chain in state x_{low}. As previously, we form the transition probability matrix Q, which we solve to give the mean number of steps until changeover (and hence the mean changeover time). As a practical note, the large values of T result in very large (but sparse) transition matrices. Using the sparse operator of the MATLAB$_{tm}$ computation package, we have been able to solve the required matrices efficiently.

5.7 Results and Discussion

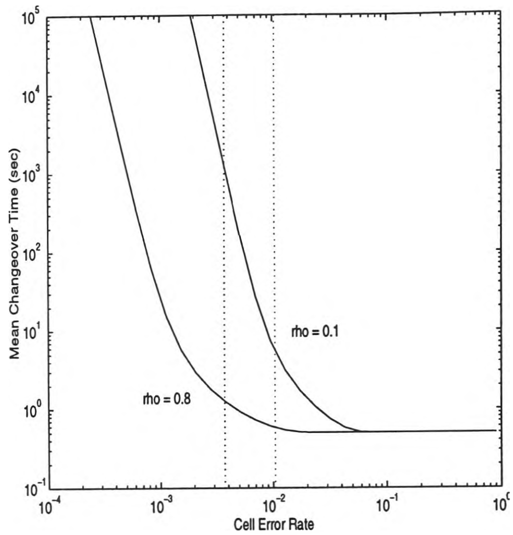
Here we present the mean changeover time versus cell error probability results for each of the three ATM error monitors. We begin by looking at error monitor performance for independent errors, then consider bursty errors.

5.7.1 Independent Errors

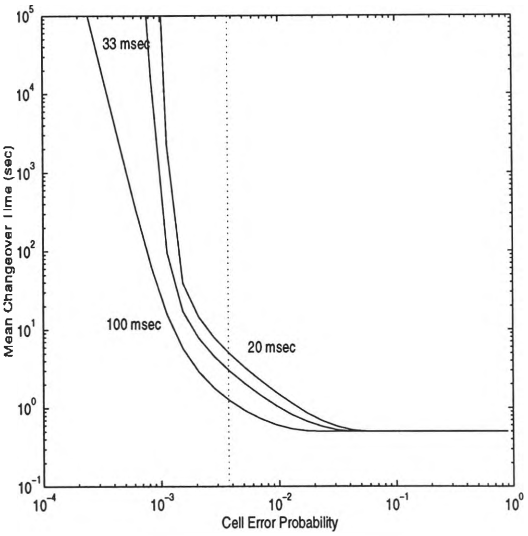
Figure 5.2a shows the EIM results for a 20 msec interval and for utilisations (marked as rho) of 0.1 and 0.8. All MSUs are 20 cells. The dotted lines show the maximum sustainable cell error rate for each utilisation, with the one furthest to the left being for 0.8. The EIM curve closely matches the changeover error threshold at a utilisation of 0.1. This is as expected, as the EIM parameters have been chosen for this utilisation. However the changeover error rate enforced by the EIM at the higher utilisation (0.8) is too conservative, as it should ideally match the dotted line as well. The inability to match the changeover error threshold at all utilisations is a key characteristic of the EIM operation in ATM environments. This is due to the different number of cells per interval at different utilisations, with the resulting change in the interval error rate. This is not a problem with the SS7 EIM, as each interval has the same number of bits.



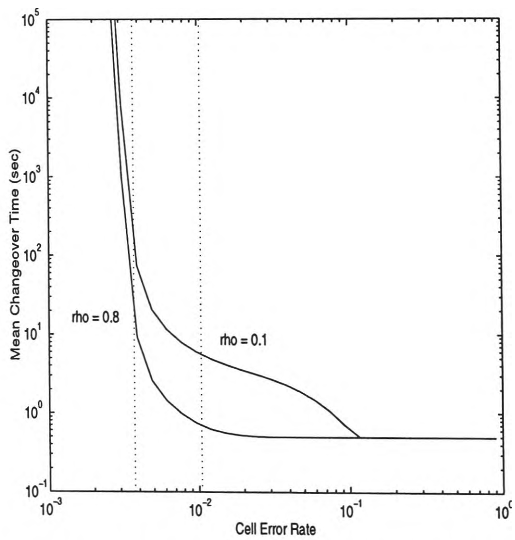
a) EIM: 5 Mbps, 20 msec interval



b) EIM: 1 Mbps, 100 msec interval



c) EIM: 1 Mbps, comparing different intervals



d) OAM: 5 Mbps, block size =189 cells

Figure 5.2 Mean changeover time vs cell error rate for EIM and OAM Error Monitors

Figure 5.2b shows the EIM results at 1 Mbps. Once again, the MSUs are all 20 cells. While the same trend is seen as for the 5 Mbps case (i.e. too conservative at the high utilisation), we also see that the changeover time vs cell error probability curve has a much wider knee. A sharper knee (i.e. like

Figure 5.2a) indicates a better error monitor, as there is less probability of spurious changeovers (i.e. at error probabilities below the changeover threshold). The problem with the 1 Mbps case is that only 5 (consecutive) errored intervals are needed to changeover, as compared to 25 intervals for the 5 Mbps example. The same problem, i.e. a wide knee, should therefore be expected on the proposed SS7 EIM, as it also requires only 5 consecutive errored intervals to changeover. This feature of the SS7 EIM has been noted previously in [KAN93.1].

Figure 5.2c investigates the EIM at 1 Mbps with shorter intervals, i.e. 33 msec (which gives 15 consecutive errored intervals before changeover) and 20 msec (which gives 25 intervals). The utilisation is 0.8, with 20 cell MSUs, as before. We see that the shorter interval produces a sharper knee, and a better error monitor. Hence when investigating the OAM monitor at 1 Mbps (e.g. Figure 5.3c), we shall choose a block size equivalent to a 20 msec EIM interval. This gives a block size of 38 cells.

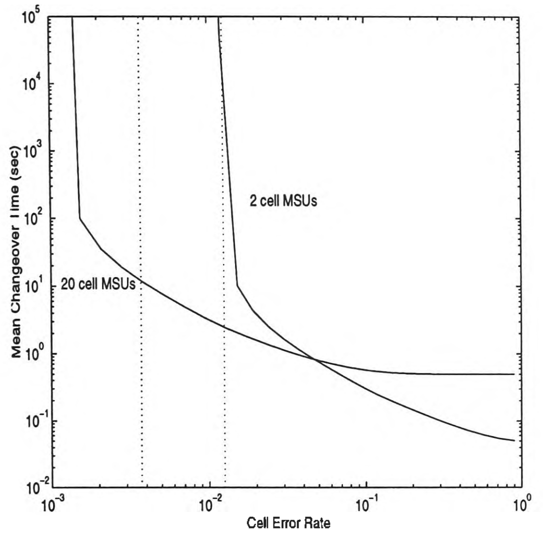
Figure 5.2d shows the results for the OAM monitor at 5 Mbps, for a block size of 189 cells (i.e. the mean number of cells in 20 msec, the EIM interval used for the 5 Mbps example). Once again, the MSUs are 20 cells, with results shown for utilisations of 0.1 and 0.8. The odd shape of the upper curve is due to the changeover time becoming constant once the effective utilisation (i.e. including retransmissions) reaches 1. While the OAM changes over at the correct cell error rate at a utilisation of 0.8 (as per the system design), the changeover error threshold at a utilisation of 0.1 is more conservative than it needs to be. This is because the OAM block error rate does not vary with the utilisation, whereas the maximum sustainable cell error rate does. The different characteristics of the EIM and the OAM error monitors, as seen in Figure 5.2a and Figure 5.2d, are because the EIM interval error rate (for a given BER) changes with the utilisation, essentially providing an adaptive scheme.

Figure 5.3a and Figure 5.3b show the SUERM performance, for the 5 Mbps and 1 Mbps links respectively. In both cases, the results show that when the link carries all 2 cell MSUs, a minimum changeover time cannot be guaranteed. This was noted in section 5.5, where we suggested using a separate timer to enforce the minimum changeover time. Clearly for the SUERM to satisfy all three requirements (i.e. those from section 5.2) in an ATM environment, this timer would be needed.

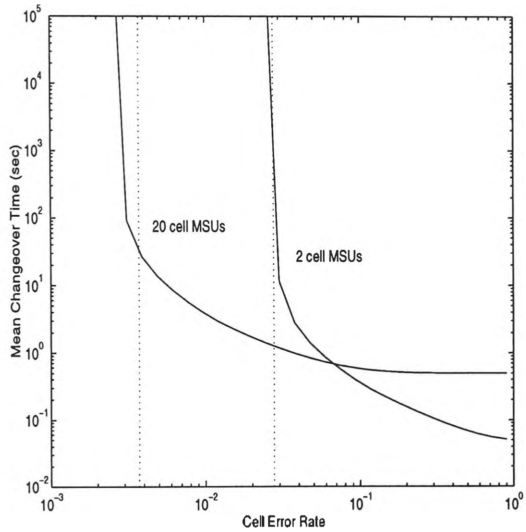
Both Figure 5.3a and Figure 5.3b show a good match with the maximum sustainable cell error rate for 2 cell MSUs, as intended by the system design. For the 5 Mbps case, the SUERM changeover error rate is conservative (i.e. less than the maximum sustainable cell error probability). However for the 1 Mbps link, the match is good.

The question is then: would the SUERM, with an additional timer to enforce the minimum changeover time, be the best choice for an ATM signalling error monitor? To investigate this, Figure 5.3c compares changeover performance, at 1 Mbps, of the OAM scheme (with a block size of 38 cells) and the SUERM (with 20 cell MSUs) at a utilisation of 0.8. As the figure shows, there is little difference in performance. Figure 5.3d shows 5 Mbps case, where a 189 cell OAM block is used. Here the OAM monitor provides better performance, as a changeover occurs more quickly once the maximum sustainable cell error rate is exceeded. For example, at a cell error probability of 7e-03 (i.e. around twice the maximum sustainable cell error probability), the SUERM takes about 5 seconds to changeover, while the OAM monitor changes over in around a second.

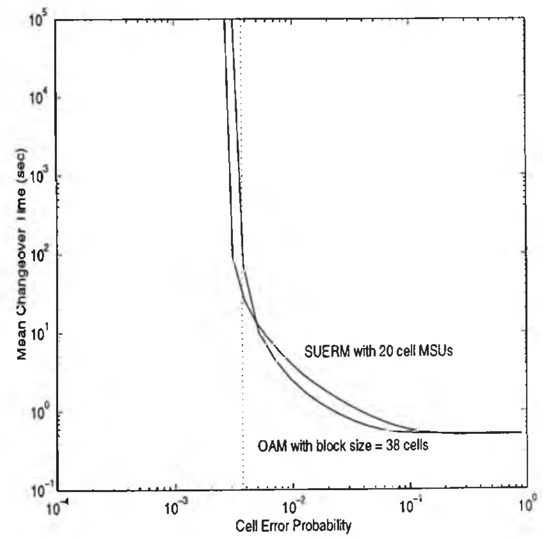
The key feature of the SUERM is that on a higher speed link (such as the 5 Mbps example), the proportionate increase in the Leaky Bucket depth increases the changeover time, thereby reducing the effectiveness of the error monitor. By contrast the OAM (or the EIM) may increase the block size



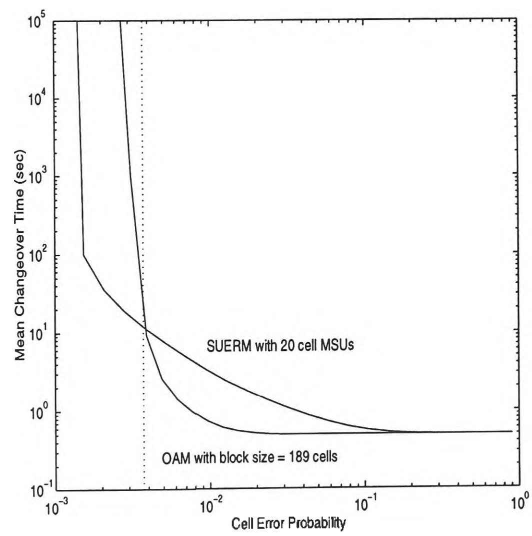
a) SUERM: 5 Mbps, utilisation = 0.8



b) SUERM: 1 Mbps, utilisation = 0.8



c) SUERM/OAM comparison, 1 Mbps



d) SUERM/OAM comparison, 5 Mbps

Figure 5.3 Mean Changeover Time vs Cell Error Rate for SUERM

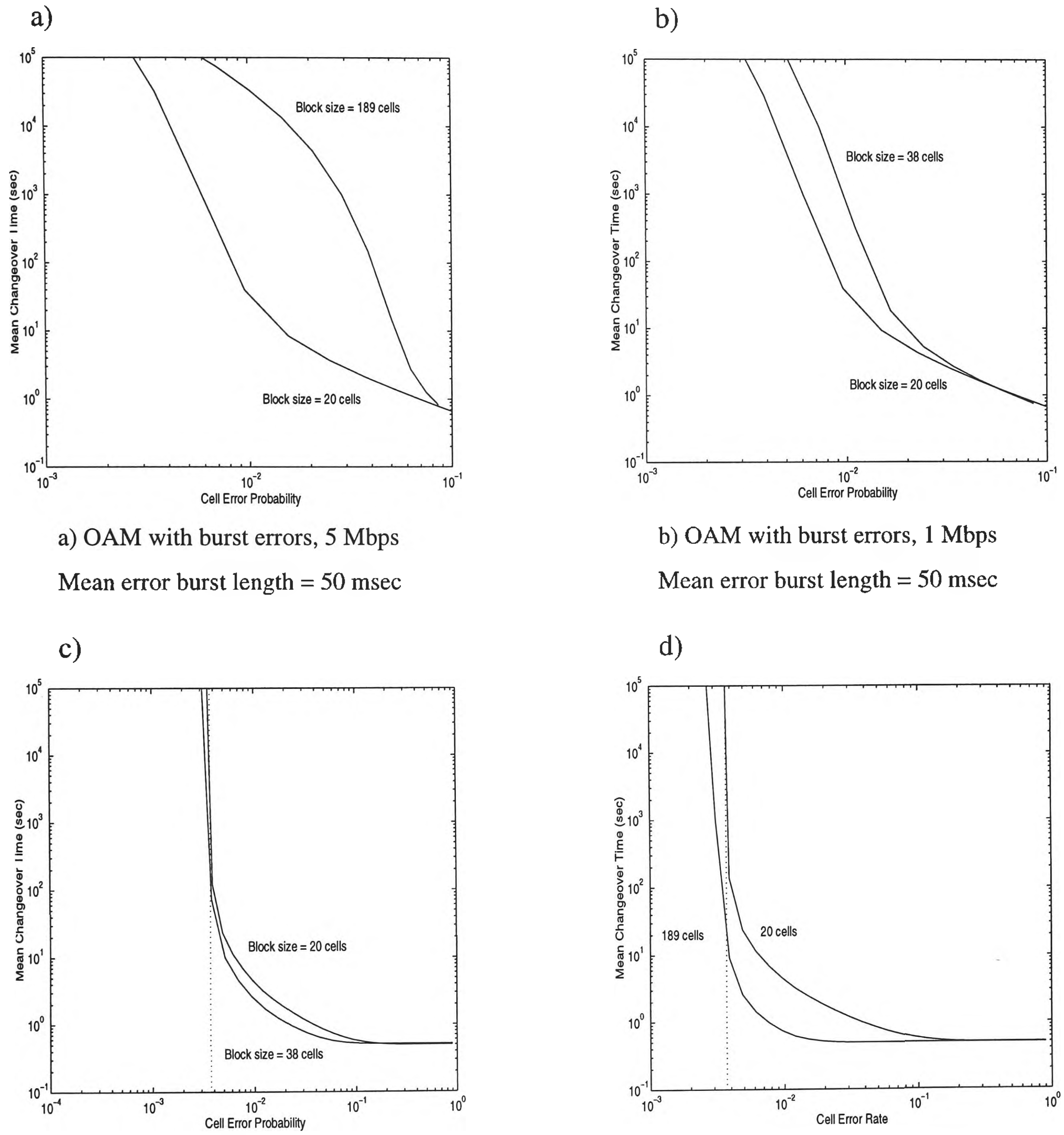
(which is not tied to the MSU size) to match the link speed, thereby providing a short changeover time. For this reason, we propose that the OAM (or possibly the EIM) schemes are a better choice than the SUERM.

5.7.2 Bursty Errors

We now consider the performance of the OAM scheme for bursty errors (we omit results for the EIM scheme here, as they are almost identical to the OAM ones). Figure 5.4a and Figure 5.4b show the changeover time vs mean cell error probability for the OAM for 5 Mbps and 1 Mbps respectively. In both cases the utilisation is 0.8 and the MSU size 20 cells. We use an error model with alternating periods of high and low error rates: the BER during the high period being 3e-04, with 1e-06 the BER during the low period (similar figures are used in the SS7 SUERM analysis of [RAM93]). The mean duration of the high error rate period is 50 msec. We arrive at the mean cell error rates on the plot by adjusting the mean low error rate period.

Figure 5.4a shows that OAM error monitor with a block size of 189 cells (and hence also the EIM with a 20 msec interval) performs poorly when bursty errors occur. For example, the link will remain in service almost indefinitely (i.e. not changeover) at a mean cell error probability of 0.02 (this corresponds to a mean low error rate period of around 250 msec). While we have not studied the effect of bursty errors on delay performance, it is reasonable to assume that a link should not remain in service at such a high error rate. When the OAM block size is reduced to 20 cells, the response to bursty errors is much better, as shown by the Figure 5.4a. In this case, at a mean cell error probability of 0.02, a changeover occurs in about 5 seconds. While a similar trend is seen in Figure 5.4b (the 1 Mbps) case, the effect is not as pronounced, as the OAM block size is reduced by a factor of about 2 only (i.e. from 38 cells to 20 cells), instead of by an order of magnitude as in the 5 Mbps case.

Hence a short OAM block (or EIM) interval, such as the 20 cell block in Figure 5.4a and Figure 5.4b, is needed to detect burst errors, particularly on high speed links (e.g. 5 Mbps). While reducing the OAM block size increases the bandwidth overhead, due to the greater number of OAM cells, the penalty in this case (i.e. an increase of 5% in signalling bandwidth) is not



a) OAM with burst errors, 5 Mbps
Mean error burst length = 50 msec

b) OAM with burst errors, 1 Mbps
Mean error burst length = 50 msec

c) OAM with independent errors, 1 Mbps

d) OAM with independent errors, 5 Mbps

Figure 5.4 Mean Changeover Time vs Cell Error Rate for OAM

likely to be significant. While the minimum OAM block size specified in ITU recommendation I.610 is 128 cells, we assume that there are no reasons why it could not be made smaller (as required here).

The need for a shorter block (or interval), to detect bursty errors, applies to the EIM as well as the OAM monitor. However reducing the EIM a similar amount (i.e. from 20 msec to 2.12 msec, the mean time for 20 cells to arrive at a utilisation of 0.8 on the 5 Mbps link) is likely to compromise the EIM performance. As indicated in section 5.5, a very short EIM interval (i.e. a few msec) is not likely to be practical. This is because the SSCOP implementation (in particular the delay jitter in reporting retransmissions to the Layer management) may affect the accuracy of the EIM error rate measurement. Hence the requirement for a short interval to detect burst errors, and the potential for inaccurate error rate measurements in an EIM implementation with such a short interval, leads us to reject the EIM as a potential ATM error monitor.

We have shown that to monitor link performance when independent errors prevail, an OAM scheme with a block size of around 200 cells provides the best performance on a 5 Mbps link (as shown by Figure 5.3d). However to detect bursty error conditions, a much smaller OAM block, of the order of the MSU size (e.g. 20 cells) is needed. It would appear then that, ideally, we should use both block sizes. This is in fact the approach we propose, i.e. to use an OAM monitor which simultaneously employs 2 block sizes. We now outline a simple extension to the OAM error monitor which we proposed in section 5.5, which allows this.

To operate with both OAM block sizes simultaneously, two counters are required, one dimensioned (using the methods of section 5.5) for the small block size, the other for the large block size. This in turn gives two sets of parameters, U_L, D_L and T_L (for the large block size) and U_S, D_S and T_S (for the small block size). The large block is chosen to be an integral number, n, of small blocks (e.g. 9 in the 5 Mbps example). The actual OAM block size used is the small one (e.g. 20 cells for the 5 Mbps example). A large "block" is deemed error free if n consecutive small blocks arrive without error. A changeover occurs if either T_S overflows (e.g. if bursty errors prevail) or if T_L overflows (e.g. if errors are not bursty). During a worst case error situation, i.e. every block errored, both counters would overflow simultaneously.

5.8 Conclusions

This chapter has achieved one of the major aims of the thesis, namely finding the best error monitor for ATM signalling links. Building upon the SSCOP delay modelling techniques of chapter 4, we have compared three potential error monitors. Our conclusion is that an ATM error monitor based on OAM cell blocks is the most suitable choice. During the course of this investigation we have made a number of observations about each of the 3 error monitoring schemes. These observations, as well as the final conclusions of this chapter, are gathered here.

The Errored Interval Monitor (EIM), proposed for high speed SS7 links, has two serious shortcomings in an ATM environment. The first is that the interval error probability changes with the link utilisation (due to the change in the number of cells per interval). This results in conservative changeover performance at high utilisations. The second shortcoming is that, due to the way that the Signalling AAL detects and reports errors, there is likely to be a lower limit on the EIM interval size. This lower limit may restrict the ability of the EIM to detect bursty error situations in an ATM environment.

Similarly the SUERM (i.e. the Leaky Bucket) has two major shortcomings in an ATM environment. Due to the variability in MSU size, it is not possible to dimension the SUERM so that it limits the changeover transient while also guaranteeing a minimum changeover time. We have proposed a method for overcoming this first problem, namely a separate timer which ensures a minimum changeover time. However the SUERM also features a much longer changeover time than the EIM or OAM monitors on high speed links, due to the limitations imposed by the MSU size.

We therefore conclude that the OAM scheme is the most suitable ATM error monitor, provided that constant block sizes can be guaranteed (the OAM standard, I.610, does not require the OAM block size to be constant). However, to be able to detect bursty errors, as well as provide a rapid changeover when non bursty errors prevail, the OAM error monitor needs to consider two block sizes (one being the actual OAM block size, the other being an integral multiple of this). We have designed a simple algorithm which allows this.

6. ATM Signalling Network Architectures and Protocols

6.1 Introduction

Previous chapters have examined the delay performance and error monitoring of ATM Signalling Links. These investigations form part of the wider question of B-ISDN MTP structure. This chapter examines this structure further, by considering ATM Signalling Network architectures and the MTP functions which they will require.

A fundamental issue is whether ATM Signalling Networks will be fully meshed (and hence use associated mode signalling only), or whether they will use the current SS7 STP based architecture. Section 6.2 examines previous studies in this area. While these studies generally favour the fully meshed approach, they present little (or no) quantitative evidence to support their conclusions. To address this, section 6.3 presents a case study which compares these two architectures from the perspective of bandwidth requirements and maximum call handing capacity. The conclusion is that the fully meshed approach should not be precluded on the basis of its bandwidth requirements. In addition, the fully meshed approach has a greater call handling capacity in many instances.

Assuming the wide use of fully meshed ATM signalling networks (or more generally, ATM signalling networks which use associated mode signalling only), the MTP protocol requirements of such networks must be examined. The remainder of the chapter examines the B-ISDN MTP, with a view to the needs of associated mode signalling.

Section 6.4 considers two options for the B-ISDN MTP3. The first option is intended for a private network using CS-1 signalling only (e.g. B-ISUP). We propose a minimal MTP3, which provides the minimum capabilities needed to transfer MSUs (thereby allowing a low cost implementation). In particular, this minimal MTP3 reuses the UNI Service Specific Coordination Function (SSCF), rather than the more complex NNI SSCF. The second option (proposed in [ATM94.1] and [ATM94.2]) results in a more complex MTP3 implementation, which provides failure recovery and maintenance features more appropriate to the needs of public networks[1].

Section 6.5 considers MTP2 functions for B-ISDN. Again we examine two options; a minimal version and a fully featured version (these align with the two MTP3 options described above). The fully featured option is essentially the one outlined in ITU recommendation Q.2140 [ITU94.5], which replaces all SS7 MTP2 functions with ATM alternatives (with the exception of the error monitoring scheme, which Q.2140 does not describe). To complement our work on ATM error monitoring, we examine the proposed ITU technique for link proving (which determines when a failed ATM Signalling link may be returned to service). We identify deficiencies with this proposed proving scheme, and outline an improved method.

1. Our work on the minimal MTP3, which appeared in [EYE93], preceded (and complements) the work of [ATM94.1] and [ATM94.2]

6.2 Previous Studies

The issue of whether ATM Signalling Networks should be fully meshed, or STP based, has been considered by a number of previous authors. Here we summarise their findings.

[SKO89] considers 4 alternatives for ATM signalling message transport. The first two involve: using the current (narrowband) SS7 network or using the ATM network to emulate SS7 circuits. Since the paper first appeared, the progress in ATM signalling standards has resolved that ATM VCCs be used to carry signalling. Hence these first two alternatives are no longer relevant. The third alternative uses ATM Permanent Virtual Channel Connections (PVCCs) within an STP network, while the fourth uses a full mesh of ATM PVCCs, without STPs. The last two alternatives, which correspond to the two architectures which we consider, are examined qualitatively in [SKO89]. The major observation is that the fully meshed approach would generate a large amount of redundant traffic (i.e. Fill In Signal Units). These are needed to monitor the performance of the large number of PVCCs associated with a fully meshed network. However the paper gives no results to accompany this observation.

Modaressi (one of the co-authors of [SKO89]) argues for the fully meshed approach in a subsequent paper [MOD91], suggesting that each pair of SPs be connected by two diverse PVCCs. Only one of these (known as the primary PVCC) is active, with the other (the secondary PVCC) remaining idle until a failure occurs. At this time a changeover is done, so that the primary and secondary PVCCs change roles.

[MOD91] dispenses with load sharing between the two links, as well as the changeback procedure described in chapter 2. Presumably if the new "primary" fails, a changeover is done back to the previous link (assuming that it has recovered). As STPs are not envisioned, the procedures for forced and controlled rerouting (described in chapter 2) are also omitted. While supporting the fully meshed approach, [MOD89] does not consider bandwidth requirements. In particular, bandwidth provisioning for the secondary PVCCs is not considered.

[GID93] suggests a fully meshed network, similar to that in [MOD91] (i.e. a pair of diverse PVCCs between each SP pair). The memory requirements to support the PVCCs in a sample 400 node network are shown to be modest (i.e. around 50 kbytes/ATM switch). [GID93] also maintains that the total bandwidth requirement for a fully meshed network is small. While a simple calculation is included to support this, no consideration is made of signalling peak rates or multiplexing techniques. The question of bandwidth requirements for fully meshed networks is considered further in section 6.3.

[FRA94] suggests that while the use of STPs in ATM signalling networks can be reduced substantially, the STP should be retained. The reason given is that "not all signalling relations can be served by associated mode signalling only" (however the authors do not provide examples to show this). Based on an assumption of a factor of 4 increase in signalling message length (the same assumption made by us in chapter 5), [FRA94] suggests a peak rate for signalling links of 500 kbps. In addition, it is suggested that ATM Signalling Links be statistically multiplexed. However there is no indication of how to determine the resulting capacity requirement, nor how the signalling network should respond to ATM cell loss in a statistically multiplexed network (due to congestion in ATM switches).

While all of these studies (except for [SKO89]) recommend the use of fully meshed ATM signalling networks, there are, so far, no quantitative analyses of capacity requirements or call handling capacity. We address these matters in the following section.

6.3 Comparison of Fully Meshed and STP Based Architectures

Here we compare the performance and capacity requirements of the fully meshed and the STP based architectures for a typical regional ATM signalling network. We begin by considering the likely signalling patterns, based on proposed B-ISDN call types.

6.3.1 B-ISDN Call Types and Signalling Messages

The initial focus on associated mode signalling for ATM is consistent with the signalling arising from B-ISDN capability set 1 (CS-1). A major characteristic of this signalling is that it occurs between adjacent SPs only. The call control protocol used for CS-1, known as B-ISUP, was described in chapter 2.

B-ISDN Capability Set 2 (CS-2) has been designed to allow more complex call types than CS-1 (e.g. point to multi-point calls, multiple connections/media per call). In addition, unlike CS-1 (and B-ISUP), CS-2 allows MSUs to be exchanged between signalling endpoints before a call is established. This feature, known as network "Look Ahead", may be used (for example) to check the availability/compatibility of a users equipment before establishing a call. The ability to provide end-to-end signalling, in addition to node by node signalling, was identified in Chapter 2 as a requirement for the B-ISDN MTP used for CS-2 signalling.

An example of a call which uses end to end signalling is given in Figure 6.1 (taken from the CS-2 draft recommendation [ITU93], p. 73). This shows the MSUs exchanged when establishing a point to point call using Look Ahead. As this call example is used in the case study which follows, we shall examine it more closely. The network signalling component of the call (i.e. our area of interest) is seen between the Requesting Serving Node, the

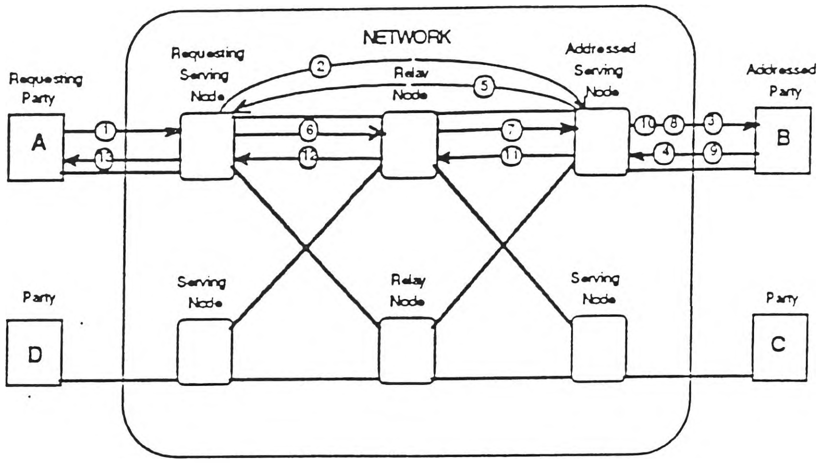


Figure 6.1 CS-2 Call Example with Lookahead

Addressed Serving Node and the Relay node. Two types of MSUs are present, the end to end ones (which shall be known as "call control" MSUs) and the node by node ones (which we shall call "bearer control" MSUs). If we assume each call to be single media/single connection, then the MSUs in Figure 6.1 have the lengths shown in Table 6.1 (based on the MSU formats given in chapter 4 of [ITU93]).

Table 6.1

MSU type	Mesg # in figure 6.1	Length (cells)	MSU name
Call control	2	6	Interrogation-Terminating-End-Point.ready
Call control	5	2	Interrogation-Terminating-End-Point.commit
Bearer control	6,7	6	Call-&-Bearer-Setup.ready
Bearer control	11,12	2	Call-&-Bearer-Setup.commit

Hence there are effectively 4 MSUs which must go end to end to establish the call (the call control MSUs go end to end directly, as shown, whilst the bearer control MSUs visit the relay node). As the MSU exchange required to clear a call is not given in [ITU93], we shall assume that it is similar to the call setup one.

6.3.2 Sample ATM Signalling Network

For this study we shall consider a typical regional network, as shown in Figure 6.2. This 38 node example (reproduced from [SHA92]), shows part of a European national network. The aim of [SHA92] is to consider alternatives for bearer locations, hence the links shown (and their respective capacities) are hypothetical. However the node locations, and the subscriber demands which have lead to the capacity choices, are real. For the purposes of our study, we shall assume that each node is a Signalling Point, and that STPs are located at nodes 14 and 5 (for the STP based option).

We shall dimension the ATM signalling network in this study to satisfy the following demands:

- Each SP generating 30 calls/sec, with these calls spread uniformly among all SPs.

- One SP generating 30 calls/sec to a single destination.

- Maximum link/STP utilisation of 0.4 (similar to that used in [SCH94]).

Using these values, we obtain the following capacity figures:

- SP to SP links: 509 kbps

-SP to STP links: 763 kbps

- STP switching capacity: 11400 MSUs/sec

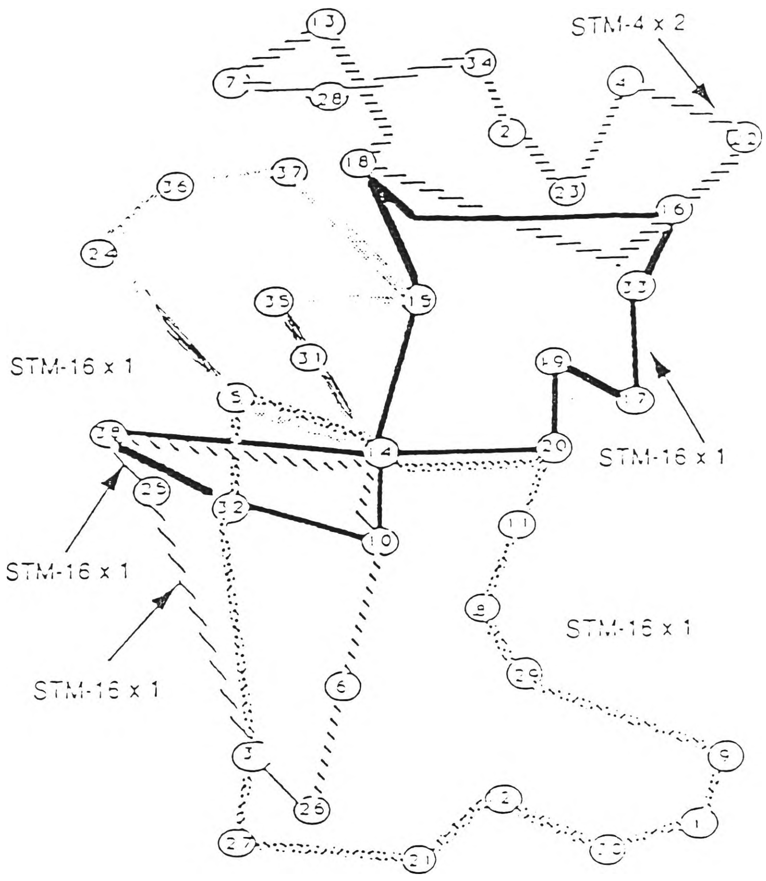


Figure 6.2 Regional Network

For the fully meshed network we shall assume two diverse links between each SP pair (note: in this context a "link" is a Virtual Channel Connection). Each SP in the STP based network maintains two diverse links, one to each STP. The route for each link is based on the shortest path.

We make some additional assumptions for this study, which are as follows:

1) All capacity is peak rate allocated only (i.e. we assume no statistical multiplexing). The reasons for this choice are:

- resulting capacity figures are worst case.

- issues regarding cell loss in ATM multiplexers do not arise

- mechanisms for establishing (and maintaining) peak rate allocated connections are already in place.

2) There is no load sharing between links (i.e. the backup link is idle for both the fully meshed and STP based networks). However the two STPs share the load equally.

3) All bearer control MSUs travel end to end for the fully meshed network, and follow the same route as the call control messages (in practice they would terminate at intermediate (i.e. transit nodes)). This ignores any savings due to statistical multiplexing of bearer control MSUs which would occur in intermediate nodes. Hence the overall capacity needed to support bearer control MSUs in the fully meshed example represents a worst case.

4) For the STP based network, we assume that all MSUs (i.e. call control and bearer control) are switched by the STPs (in practice, associated mode signalling would probably be used for some of the bearer control messages).

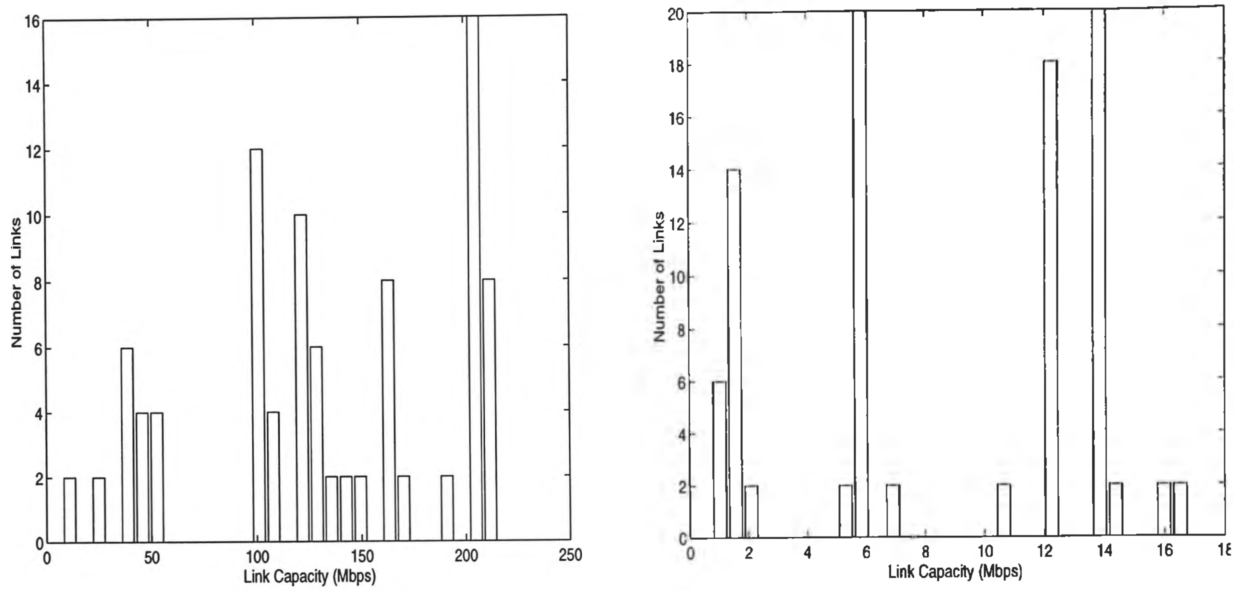
5) All ATM signalling links are PVCCs (as suggested in [SKO89], [MOD91] and [GID93]). While Switched VCCs may be used in principle, we reject this possibility for the following reasons:

- If a backup SVCC is set up ahead of time, then a separate routing protocol will need to be devised to ensure that the backup SVCC takes a diverse route to the main SVCC.

- If backup SVCCs are setup only when needed (i.e. when a changeover occurs), then the call processing load to simultaneously re-establish a (potentially) large portion of the signalling network may disrupt regular call setup. This would apply particularly for the fully meshed case.

6.3.3 Results

In this section we consider the capacity requirements for the sample network, as well as the respective performance of each architecture during overload.



a) Fully Meshed Network b) STP Based Network

Figure 6.3 Histograms of link capacities for Fully Meshed and STP Based ATM signalling networks

Capacity Requirements

Figure 6.3a and Figure 6.3b show histograms of the aggregate link capacities for the fully meshed and the STP networks respectively. These are obtained by summing the capacities of the PVCC segments which pass through each link (a C program has been developed to produce these results. This program determines the shortest path for all OD pairs, then finds the resulting link capacities). The higher capacity links are those in the middle of the network, which carry more transit PVCCs. Clearly the link capacities for the STP network, which have a mean of 8.56 Mbps, are an insignificant proportion of the total network capacity (while the PVCCs in the STP network have a greater capacity than those in the fully meshed network (763 kbps as opposed to 509 kbps), the STP network has far fewer PVCCs. Hence the aggregate link capacity in the STP network is much less than that for the fully meshed network). The fully meshed network, with a mean link capacity of 133.10 Mbps, uses far more bandwidth. However this is still only a few percent of the total link capacities of the network shown in Figure 6.2 (an STM-16 link provides 2.4 Gbps, while the combination of 2xSTM-4 provides 1244 Mbps). Of course, the increased bandwidth cost of the fully meshed network is offset by the savings in STPs.

Call Handling Capacity

We now consider the respective call handling capacities of the two network architectures during overload. To do this, we consider end to end MSU delay performance (minus propagation delay). For the fully meshed case, this is simply the PVCC (i.e. SSCOP) transmission delay, as given in chapter 4 (we assume a poll rate of 10/sec and an error free link). For the STP case, we arrive at the delay by modelling the MTP3 processor as a single M/G/1 queue, with a service rate equal to the STP capacity obtained earlier (i.e. 11400 MSUs/sec). A similar STP model was used in [RUM94.1] where a simulation was used to capture the effect of repeated call attempts and failed call releases. The total end to end delay is the sum of the MTP3 delay and the delays in the incoming and outgoing PVCCs.

Figure 6.4a shows MSU delay vs call handling capacity per SP, when every SP in the network generates calls at the same rate (and where calls destinations are distributed evenly through the network). The fully meshed network has a far greater call handling capacity than its STP based counterpart. This is due to two factors: the increased bandwidth allocated to the fully meshed, network, as well as the absence of a centralised bottleneck (i.e. an STP). We note however that the maximum calling rates for the fully meshed case (greater than 1000 calls/sec) could never be achieved in practice, as the call setup processes in the transit nodes would overload long before this rate was reached.

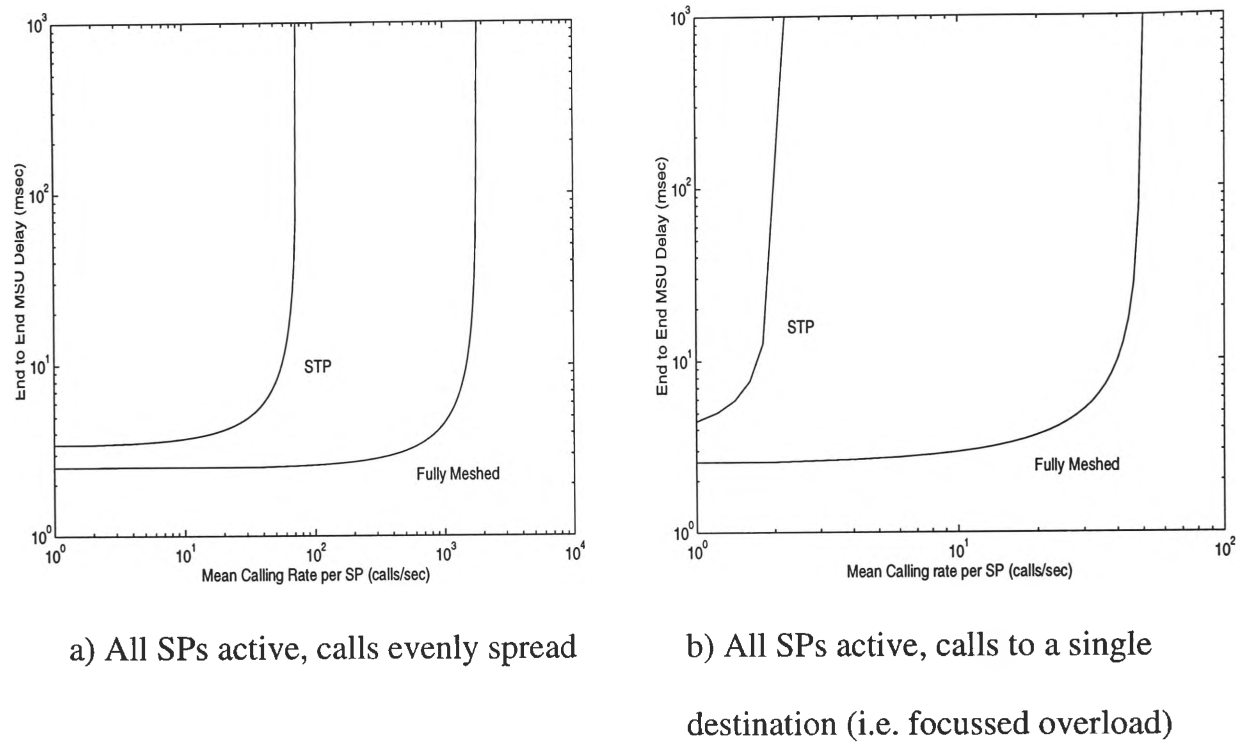


Figure 6.4 Comparison of Delay vs Calling Rate for Fully Meshed and STP Based Networks

Figure 6.4b shows the MSU delay performance when all SPs generate the same number of calls, all of which have the same destination (once again, background traffic (i.e. to other destinations) is ignored). This is the focused overload situation considered in [RUM94], which, for example would be typical of a media stimulated calling event. In this instance the fully meshed network provides the best performance.The main observation here is not the number of calls carried (as most would be rejected as the destination becomes overloaded), but rather that the fully meshed network is able to restrict the overload to only those PVCCs which go to the affected destination. By contrast, the same scenario in the STP case may degrade the performance of the entire network, particularly if level 3 of the STP becomes overloaded. The reason for this is that congestion at level 3 of an STP causes all MSUs to be delayed, not just those to the overloaded destination.

The purpose of this study has been to examine the capacity requirements and call handling capability of a fully meshed ATM signalling network, as these areas appear not to have been considered previously. Our results show that, for a typical regional network, the fully meshed approach uses a modest fraction (i.e. less than 10%) of the total bandwidth. Hence bandwidth considerations are not likely to preclude the deployment of fully meshed networks, even for the worst case which we have considered here (i.e. peak rate allocation). While we have considered peak rate allocation only, for the reasons mentioned above, signalling traffic may be statistically multiplexed (as suggested in [FRA94], [GID93]). Hence the B-ISDN MTP protocols should allow for statistical multiplexing. We shall keep this requirement in mind when examining the proposed ITU replacement for MTP2.

This study has considered only the internal traffic within a regional network. While the fully meshed network in this example could be expanded to include further nodes (e.g. those from other networks), there is clearly an upper limit to the fully meshed network size, due to capacity constraints. In addition, different networks may not be prepared to allow their respective SPs to communicate directly (this is the case with current SS7 networks [GOL90])). Hence it is likely that STPs will still be deployed (as suggested in [FRA94]), to limit the size of individual networks and to perform gateway screening between networks. However these STPs will carry a much smaller proportion of the ATM signalling traffic than is presently the case for SS7 STPs.

6.4 MTP3 Requirements for ATM Signalling

Here we consider the MTP3 for ATM Signalling networks, with an emphasis on the needs for associated mode signalling. We identify two types of ATM signalling networks, each having different MTP3 requirements. These two network types are as follows:

i) Private networks which use CS-1 signalling only (e.g. B-ISUP or Q.2931), and where signalling links follows the same paths as the VCCs which they establish. Here signalling messages travel between adjacent SPs only (i.e. while associated mode signalling is used, the network is not fully meshed). We propose a minimal MTP3 for this network type, which provides just the minimum capabilities needed to transfer MSUs (thereby allowing a low cost implementation).

ii) Public networks, where the required failure recovery and maintenance procedures result in a more complex MTP3 implementation. While still using only associated mode signalling, such a network may be fully meshed (like the one in our case study). This is to fulfil the CS-2 requirement for end to end MSU transfer (i.e. for call control signalling).

We begin by describing the minimal MTP3.

6.4.1 Minimal MTP3

The MTP3 for both network types will sit above the Signalling AAL. As Figure 6.5 shows, the Signalling AAL supports both UNI and NNI signalling. This is done by combining common protocol elements (i.e. the AAL5 and the SSCOP) and dedicated protocol elements, namely the Service Specific Coordination Functions (SSCFs). The NNI SSCF proposed in ITU recommendation Q.2140 interfaces to MTP3, and supports a set of functions equivalent to those of the SS7 MTP3.

As the private network which we consider will use associated mode signalling only, most of the SS7 MTP3 functions will not be required (as they are in place to support STP operation). Accordingly the SSCF at the NNI, which interfaces with our proposed minimal MTP3, may be simplified. An obvious candidate for a simplified SSCF is that used at the UNI[2] (and described in ITU recommendation Q.2130). This merely passes PDUs (and requests for link establishment/release) between the SSCOP and the Layer 3 signalling

2. Reusing the UNI SSCF at the NNI was initially suggested by [PRI93]

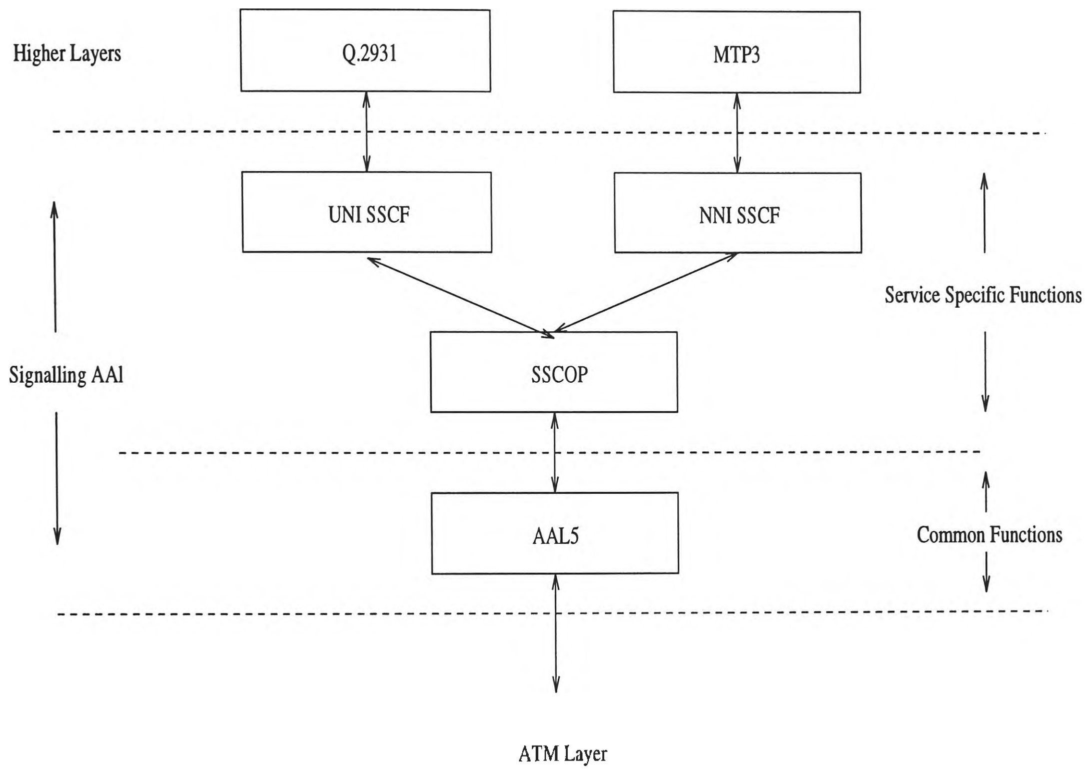


Figure 6.5 Signalling AAL

protocol (Q.2931). We now describe a minimal MTP3, for private ATM signalling networks, which uses this proposed Signalling AAL.

The SS7 MTP3 functions fall into two major categories: signalling message handling (i.e. message discrimination, distribution and routing) and signalling network management (link changeover/changeback, congestion control etc.). We consider first the signalling message handling functions needed for a minimal ATM MTP3. As an SP may have signalling links to a number of adjacent SPs, the message routing capability (which associates a Point Code with an outgoing link) will need to be kept. Message discrimination identifies mis-routed messages, by ensuring that the Point Codes of arriving MSUs match the SPs own Point Code. If the Point Codes do not match, the SP discards the MSU. Hence message discrimination in an SP provides a maintenance function (i.e. checking for mis-routed MSUs) which increases robustness. We therefore recommend that the message discrimination function be kept. The message distribution function, which determines the target

User Part within an SP, will however not be needed. This is because the SPs in the private network considered here would have a single User Part only, either B-ISUP or Q.2931 (not both).

The SS7 MTP3 Signalling Network Management functions are designed to reconfigure signalling networks in the event of failure or congestion, and to allow signalling network testing. In particular, the failure recovery functions prevent MSU loss/duplication/mis-sequencing in SS7 signalling link sets (i.e. multiple links in parallel) and SS7 signalling routes (i.e. multiple links in tandem). An associated mode network however would not have signalling routes, as there are no STPs. Hence all of the MTP3 management functions pertaining to signalling routes, such as the forced rerouting and time controlled diversion procedures described in chapter 2, would be omitted from the minimal MTP3. In accordance with our aim of providing basic functionality only, the MTP3 maintenance functions (e.g. signalling link test, signalling link restart, management inhibiting) would also be omitted.

The two remaining SS7 MTP3 Signalling Network Management functions are congestion control and link changeover/changeback. We now examine whether they are needed in the minimal MTP3.

Congestion Control and Changeover/Changeback

SS7 congestion control is invoked by STPs, which send MTP3 "Transfer Controlled" (TFC) messages to the SPs which are generating the excess traffic (this procedure was described in chapter 2). As STPs will not exist in the signalling network considered here, TFC messages will not be present. Hence the SS7 congestion control procedures based on the TFC message, would also not exist in the minimal MTP3.

The minimal MTP3 would however still provide congestion control locally within an SP. This would be invoked when the transmit queue of an outgoing link exceeded a set threshold (methods for determining this threshold would be those outlined in the studies on SS7 congestion presented in chapter 2). The congestion control scheme would be similar to that described in Q.704 (section 11.2.3.1, item (i)), where a congestion indication is sent to MTP3 every nth message (n=8) while the queue length exceeds this threshold. The minimal MTP3 would in turn inform B-ISUP of the congested destination, using the equivalent of the SS7 MTP3 STATUS primitive (with a cause of "signalling network congested"). Upon receiving this primitive, the B-ISUP traffic reduction procedure (outlined in Q.2764) would be invoked.

Changeover, changeback and load sharing will be needed only if multiple ATM signalling links are used in parallel. Hence we must determine whether this would be done in the private networks considered here (note that our case study, which considered a public network, used multiple signalling links between SP pairs). We begin by looking at how SS7 link sets are used.

SS7 signalling link sets serve two purposes: they provide increased capacity (i.e. more links) on a signalling route and provide alternate paths when links fail. In an ATM signalling network however, the required signalling capacity could be provided by a single VCC. However it may be desirable to provide multiple signalling links between SP pairs, for reliability reasons. For the private networks considered here, a signalling link would be the default signalling VC within a VP. Hence a signalling link failure would (in all likelihood) mean that this VP (i.e. between the two SPs) had failed as well. An alternate signalling link (if deployed) would lie within another VP, on a diverse route, and would set up user VCCs within this other VP. In addition to providing a backup in case of failure, this alternate signalling link/VP may be in place to allow users to access capacity over two separate routes. We therefore conclude that the minimal MTP3 should support alternate signalling links. Hence the load sharing and changeover/changeback functions will need to be kept.

As described in chapter 2, SS7 load sharing is done using the SLS field in the MSU header. We propose the same scheme for the minimal MTP3. SS7 changeover/changeback is coordinated by the exchange of MTP3 changeover and changeback orders. However, for the minimal MTP3, a signalling link failure will instead be indicated by the release of the SSCOP connection (as will be described in the next section). When the minimal MTP3 becomes aware of the failed link, through the arrival of a AAL-RELEASE primitive from the (UNI) SSCF, it would adjust its routing table to divert traffic to the alternate link. If an alternate link is not present, B-ISUP would instead be informed of the unavailable destination (the SS7 MTP3 has a primitive which does this, i.e. the MTP-PAUSE primitive. This primitive would be retained for the minimal MTP3).

Once a link has failed, the minimal MTP3 would initiate a simple alignment procedure, consisting of periodic attempts (e.g. every 5 seconds) to re-establish the SSCOP connection on the failed link (a successful re-establishment would imply link recovery). A changeback would be done after a successful SSCOP re-connection attempt, and would simply be the reverse of the changeover (i.e. restoring the routing table or sending an MTP3 primitive to B-ISUP indicating an available destination (i.e. the MTP-RESUME primitive)).

The SS7 changeover/changeback provides message retrieval during changeover, and ensures that message mis-sequencing does not occur during changeback. The changeover/changeback scheme proposed here for the minimal MTP3, while far simpler than its SS7 counterpart, does not provide these two functions (however B-ISUP would be able to recover from potential message loss/re-sequencing, at the expense of losing some calls). Omitting these two capabilities is consistent with our aim of providing a basic set of functions while minimising implementation cost. However to avoid repeated changeover/changebacks, the simplified alignment procedure would retire a link from service (i.e. cease automatic alignment attempts and call for manual intervention) if it: a) experienced more than a specified number of changeovers (e.g. 4) within a set period (e.g. 30 minutes) or b) could not re-establish the SSCOP connection within 10 minutes (note: similar conditions are placed on failed SS7 links).

The minimal MTP3 presented here provides the basic functions of MSU transfer, link changeover/changeback, load sharing and congestion control. We have described an implementation which is substantially simpler than its SS7 MTP3 counterpart (which includes a variety of maintenance functions, in addition to the basic functions described above). The minimal MTP3 uses no management messages, and requires only the following functions:

- message routing and discrimination (and the ability to change routing dynamically should a link failure occur).

- a rudimentary link alignment procedure (i.e. the SSCOP link establishment procedure).

- the current SS7 MTP3 primitives indicating unavailable (or congested) destinations.

In addition, this minimal MTP3 uses the UNI SSCF, which is substantially simpler than the NNI SSCF described in Q.2140.

6.4.2 Fully Featured MTP3

Here we describe the proposals made in [ATM94.1] and [ATM94.2] for an ATM MTP3 which supports associated mode signalling only. In addition to the basic functions of the minimal MTP3, the more fully featured MTP3 proposed by these authors provides additional maintenance capabilities. These would be consistent with the requirements of a public ATM signalling network (i.e. the second network type described at the start of this section). It should be noted that the composition of the ATM MTP3 is still a matter for discussion, hence the proposals both in this thesis and in [ATM94.1] and [ATM94.2]. However these proposals indicate a clear trend for a reduced featured MTP3 (compared to the SS7 version) for use in associated mode ATM signalling networks.

The MTP3 proposals of [ATM94.1] and [ATM94.2] both outline a similar set of features. The NNI SSCF (i.e. the one described in Q.2140) is used, as well as MTP3 management messages. Similar to our minimal MTP3, these two MTP3 proposals omit all functions pertaining to STPs (i.e. the route management functions). Here we list the functions from the SS7 MTP3 which [ATM94.1] and [ATM94.2] have used.

Signalling Message Handling: In addition to message routing, both proposals include message discrimination (to detect mis-routed messages) and message distribution (to allow more than one user part within an SP). The minimal MTP3 omits the latter functions, for the reasons described previously.

Changeover/changeback: The SS7 MTP3 changeover/changeback procedures are retained. These are more complex than the ones proposed for the minimal MTP3, as they provide message retrieval during changeover and prevent message mis-sequencing during changeback. In addition an error monitor which measures error rate (such as the one suggested in chapter 5) would be used, rather than the simple test for link failure done by the minimal MTP3. This increased functionality, at a cost of additional complexity (compared to the minimal MTP3) is consistent with the more stringent demands of a public network with regard to message loss and signalling delay.

Congestion Control: Both proposals recommend keeping congestion control procedures. While [ATM94.2] does not specify what they should be, [ATM94.1] indicates that only local congestion controls (i.e. within a congested SP) should be used. These local congestion controls are similar to he ones we have proposed for the minimal MTP3.

Signalling Link Management: The two proposals include the Signalling Link activation, restoration, deactivation and testing procedures. We briefly compare (and contrast) these functions with those of the minimal MTP3.

- SS7 Signalling link activation requires a link to first pass an initial alignment test, then pass a signalling link test procedure. The latter involves an exchange of MTP3 management messages, which ensure that each end a) responds to the correct point code and b) can send a given test pattern. Essentially link alignment is a test of MTP2 (or the ATM equivalent), while the signalling link test checks the correct operation of MTP3. While this degree of testing may be appropriate for a public network, it is not (in our view) required for a private network. Hence the minimal MTP3 "link activation" is just the establishment an SSCOP connection on the default signalling VC within a VP.

- Signalling link restoration is the repeated application of the link alignment procedure, until either the link re-enters service or manual intervention is sought. We have included this feature in the minimal MTP3 (with a simplified link proving procedure).

- Signalling link deactivation is simply the release of a signalling link.

Signalling Point Restart: This delays the start of signalling link operation while SPs exchange routing information. We have not included this in the minimal MTP3, as each signalling link goes to a single destination (SP) only (in the SS7 case, a single link to an STP may access many destinations. The Signalling Point Restart procedure allows time for information regarding the unavailability of any of these destinations to be transferred).

This section has examined two ATM MTP3 implementations. The first (proposed by us) provides minimum functionality, at minimum cost, in keeping with the requirements of private networks. The second MTP3 implementation, which has been proposed by others, provides the increased robustness required for public networks. We now consider the ATM "MTP2" requirements which correspond to these two MTP3 implementations.

6.5 MTP2 Requirements for ATM Signalling

Chapter 2 outlined the level 2 (i.e. MTP2) functions in SS7. This section describes how these functions may be implemented in an ATM environment. We consider two approaches. The first provides a simple MTP2 equivalent, to be used in conjunction with the minimal MTP3 of the previous section. The second approach (adopted by the ITU and outlined in Q.2140), is designed to interface to the current MTP3 (i.e. with all STP functions included), as well as the ATM MTP3 proposals of [ATM94.1] and [ATM94.2]. The additional features included in this second approach provide increased robustness, at the cost of a more complex implementation. In keeping with the major theme of this thesis, i.e. link failure detection (and recovery) for ATM signalling, we examine (and propose an improved version of) the ATM signalling link proving scheme described in Q.2140.

6.5.1 Simplified ATM MTP2

The previous section proposed a minimal MTP3, which would be served by the UNI Service Specific Coordination Function (SSCF), instead of the more complex NNI SSCF. Here we consider the provision of MTP2 capabilities in an ATM environment, based on this use of the UNI SSCF.

The provision of MTP2 functions for ATM signalling has followed developments in ATM standards, in particular the introduction of AAL5 and the SSCOP. These two components of the Signalling AAL, shown in Figure 6.5, replace the key MTP2 functions of delimiting signal units, error correction and flow control. The SSCF provides the remaining MTP2 functions, which comprise link alignment, remote processor outage indication and link error monitoring. We consider each of these separately.

The link alignment procedure ensures that the error rate on a new signalling link is below a given level before traffic is admitted. Details of the SS7 link alignment scheme, and the accompanying link proving algorithm, were given in chapter 2. The simplified MTP2 proposed here employs the SSCOP link establishment procedure as a substitute for the normal MTP2 link alignment procedure. While SSCOP link establishment provides no guarantee of error performance, the resulting message exchange (i.e. a BEGIN and BEGIN-ACKNOWLEDGE PDU) ensures that AAL5, the ATM layer and the physical link are operating before MSUs are accepted for transmission.

MTP2 uses dedicated status messages, known as Link Status Signal Units (LSSUs), to announce changes in link state. The aim of the simplified MTP2 is to eliminate these messages, and instead use the SSCOP management capabilities to perform the same task. With this approach in mind, we now consider the remaining MTP2 requirements, i.e. remote processor outage indication and error monitoring.

Remote processor outage refers to a failure above level 2. During this condition the link operates normally, however arriving MSUs cannot be processed. As there is no point in sending MSUs at this time, the simplified MTP2 at the affected end would release the SSCOP connection. Once the condition had cleared, the affected SP would re-establish the SSCOP connection. Any connection establishment attempts arriving from the remote SP while the processor outage was current would be rejected.

Link error monitoring in the simplified MTP2 would also be provided by the SSCOP. Here the keep-alive feature provided by the poll and status messages (which we described in chapter 4) would be used. Similar to our proposed link alignment procedure, this detects complete link failures only, and does not monitor the link error rate (like the schemes outlined in chapter 5 do). The SSCOP releases a connection when a link failure is detected (this is a standard feature of the SSCOP protocol). In the simplified MTP2, this would result in the loss of the MSUs queued for transmission.

6.5.2 ATM MTP2 Proposed by the ITU

The previous sub-section described a simplified MTP2 implementation, which uses the UNI SSCF described in ITU recommendation Q.2130. Here we describe the SSCF proposed by the ITU for the NNI (and its accompanying MTP2 features). This SSCF is described in ITU recommendations Q.2140 and Q.lm-nni[3], and provides greater robustness than the simplified MTP2 (with an accompanying increase in implementation cost). In particular it provides error monitoring, message retrieval after link failure and link proving. Here we highlight the protocol differences between Q.2140 and the simplified MTP2. A detailed analysis of the proposed ITU link proving scheme then follows, in sub-section 6.5.3.

Q.2140 re-introduces the LSSU, which it calls an SSCF PDU (this was omitted from the simplified MTP2 proposal outlined above). These SSCF PDUs are identified by their length, which is 4 octets (all MSUs are longer than this, and may be identified thus). They are sent either as normal SSCOP sequenced data PDUs, or in the User-to-User information field of SSCOP connection establishment/release messages.

Similar to the proposed simplified MTP2, Q.2140 releases an SSCOP connection when processor outage occurs. The SSCOP END message which is used to do this carries an SSCF PDU, indicating the processor outage status.

While Q.2140 specifies ATM signalling error monitoring, an error monitoring algorithm is not prescribed (hence our proposal in chapter 5). When a link changeover occurs, Q.2140 retrieves unacknowledged MSUs from the SSCOP and passes them to MTP3. These MSUs are then transferred to the new link (the SS7 procedure to determine which messages should be retrieved was described in chapter 2). As Q.2140 would use the current MTP3 (or the associated mode version proposed in [ATM94.1] and [ATM94.2]), the SS7 procedure would presumably be used for ATM as well.

3. Note that our work on the simplified MTP2, which was first described in [EYE93], was done before Q.2140 was produced.

6.5.3 ITU Link Proving

The SS7 link proving algorithm, known as the Alignment Error Rate Monitor (AERM), is invoked by the link alignment procedure. The SS7 AERM uses test traffic, comprising LSSUs, to measure the error rate of a signalling link (as described in chapter 2). This error rate must be less than a specified figure before the link may go into service. Link alignment is done either when a link comes into service for the first time, or after a changeover (to determine when the failed link may be brought back into service).

[MOD91] and [GID93] both question implicitly the need for link proving, as their ATM signalling proposals omit the changeback procedure (link proving determines when a changeback should occur). It is our view however that link proving is needed, even if changebacks are not done. Our reasoning is as follows. If two diverse links are provided between SPs, and one fails (resulting in a changeover), then a new backup is needed. There are two options for this, either re-using the original failed link when it recovers (as is done in SS7), or establishing a new link. The route followed by this new link would need to avoid the active route, as well as the failed portion(s) of the original route. Clearly the simplest of these two options is the first, i.e. restoring the failed route (the second option would be used if the failed route does not recover within a set time). This two tiered approach is used in SS7 [RAM93], which attempts to restore a failed link automatically, then calls for manual intervention if the restoration does not succeed. To employ this two tiered approach in an ATM signalling network, link proving would be needed.

We now describe the link proving procedure proposed in Q.2140, then examine its performance. Q.2140 outlines two proving procedures, emergency and normal. Emergency proving, which is used when a link needs to go into service as quickly as possible, consists of the SSCOP link establishment procedure only. This is the same as our proposed link proving scheme for the simplified MTP2. The normal proving procedure establishes the SSCOP connection, then sends SSCF PDUs at regular intervals, which are set by the timer T3 (each SSCF PDU occupies a single cell). The proving is successful if no errors are reported after 1000 PDUs are sent (Q.2140 is actually ambiguous as to whether any errors are allowed among the 1000 cells. We assume that they must be error free). The load resulting from the proving traffic is prescribed to be "half the nominal rate of the signalling link". As "nominal rate" is not defined in Q.2140, we shall assume it to be the peak rate. Q.2140 states that, while the link proving parameters which it supplies assume a 64 kbps peak rate, they "provide satisfactory performance over a wider range of operating environments". This assertion, which implies that the proving scheme will work for any peak rate, will be tested shortly.

If a proving attempt fails, the SSCOP connection is released. After 5 seconds, another proving attempt begins. If successful proving does not occur after 30 seconds, then the link is returned to the out of service state. This is similar to the end of an SS7 link alignment attempt (which also comprises a number of proving attempts). Q.2140 does not indicate what happens once the link is out of service. We shall therefore assume that the procedure is the same as the one outlined in [RAM93] (which applies to US SS7 links), where a failed link alignment attempt is immediately followed by another. If the SS7 link has not returned to service after 8-10 minutes, further alignment attempts are suspended, and manual intervention is flagged.

6.5.4 Deficiencies with the ITU Link Proving Scheme

We now examine the proving scheme proposed in Q.2140 from two perspectives; the impact which the resulting test traffic has on signalling network performance, and how well the scheme tests for link functionality. We identify (and provides solutions for) a number of deficiencies of the proving scheme of Q.2140, as well as proposing some enhancements.

Effects Caused by Test Traffic

Q.2140 specifies that testing traffic be sent at fixed (i.e. deterministic) intervals, so that the resulting load is half the "nominal" signalling rate (which we interpret as being half the peak rate, as mentioned previously). Such a traffic load is not likely to cause problems in a fully meshed network which uses peak rate allocation, such as the one described in section 6.3, as the bandwidth has already been provisioned. However if the signalling traffic is to be statistically multiplexed (as suggested in [FRA94] and [GID93]), then the ITU link proving scheme would cause two potential problems. We outline each of these in turn, and propose solutions.:

Burstiness: To obtain significant capacity gains from statistical multiplexing, the burstiness (i.e. ratio of peak rate to average rate) should be greater than 10 [ATK92]. This value for burstiness would be typical for the signalling VCCs in a fully meshed network, as each would normally carry only a small proportion of the total traffic generated by an SP. However a signalling VCC carrying the ITU test traffic has instead a burstiness of 2, thereby resulting in virtually no capacity savings through statistical multiplexing (i.e. to obtain an acceptable cell loss rate, the capacity would be about the same as for peak rate allocation). As all of the signalling VCCs on a given bearer could potentially carry test traffic at the same time (e.g. if an adjacent bearer fails), the signalling bandwidth would need to be sufficient to carry this load If this were not so, the test traffic would, in itself, cause cell loss in ATM multiplexers. This in turn would cause proving attempts to fail on a potentially good link.

Hence a fully meshed, statistically multiplexed ATM signalling network would need to be dimensioned to support the link proving algorithm in Q.2140, rather than the normal signalling traffic load (clearly an unsatisfactory situation). There is hence a trade-off between using a realistic traffic load for link testing and providing an opportunity for statistical multiplexing on high speed links (i.e. peak rates in the order of Mbps). We therefore propose that a link utilisation of 0.1 be used instead of the ITU figure of 0.5. This lower figure, while still representing a significant traffic load, corresponds to a burstiness of 10. As mentioned above, this allows for significant statistical multiplexing gains.[4]

Independence: Procedures for determining capacity on statistically multiplexed links assume independent sources [HUG91]. However this will not be the case for VCCs carrying ITU test traffic, as they all have the same interval between cell emissions (i.e. T3). Hence if the test traffic causes cell loss (for example within a statistically multiplexed VP), then a similar cell loss will occur after T3 seconds (as the same number of sources will again be active) and so on. To avoid this problem, T3 should be randomised. A simple solution would be to calculate a pseudo-random number T3' after each cell (i.e. SSCF PDU transmission), where T3' is uniformly distributed between 0 and 2T3. T3' would then be the interval until the next SSCF PDU transmission. This would maintain the same average rate as the previous (i.e. fixed T3) approach, while avoiding potential problems due to synchronised sources.

As mentioned, these problems would not arise (and hence the proposed solutions would not be necessary) in a peak rate allocated signalling network. We have outlined these two problems to make the point that the proposed ITU proving scheme precludes statistical multiplexing. Should statistical multiplexing therefore be required, the two changes noted above (i.e. reduction of the utilisation of links under test and the randomisation of T3) would be needed.

Testing Link Functionality

Here we examine how well the ITU proving scheme tests link functionality. We begin by reviewing the SS7 link proving scheme (i.e. the Alignment Error Rate Monitor (AERM)).

4. Note that for 64 kbps links, a utilisation of 0.5 would be required to avoid excessive proving period lengths (e.g. it would take more than a minute to pass 1000 cells on a 64 kbps link with a utilisation of 0.1, as opposed to 13.25 seconds when the utilisation is 0.5). While a utilisation of 0.5 would preclude statistical multiplexing (for the reasons outlined), 64kbps ATM signalling networks are likely to use peak rate allocation anyway (since the bandwidth requirements arising from peak rate allocation will be modest compared to those for higher speed (i.e. Mbps) links).

[RAM93] identifies a shortcoming with the way the SS7 SUERM/AERM schemes work together. The results in [RAM93] show that a link with a BER in the range $2x10^{-5}$ to $1x10^{-4}$ will be failed by the SUERM, then brought back into service by the AERM, failed once more by the SUERM, with the process repeating ad infinitum. The resulting oscillatory behaviour, which had been observed in the field (and which prompted the work in [RAM93]), clearly disrupts signalling operations. [RAM93] identifies the problem to be poor matching of the SUERM and the AERM parameters: essentially the SUERM will fail a link (with high probability) when the BER is in the above range, while the AERM will pass the link (with high probability). We now consider (from this perspective) the combined performance of the ITU proving scheme and the error monitors proposed the chapter 5.

As in chapter 5, we shall consider results for 1 Mbps and 5 Mbps links (note: the 5 Mbps link has a far greater capacity than that the links in the case study of section 6.3 (i.e 508 kbps and 763 kbps). A link of this size could, for example, carry the signalling traffic to a heavily used network resource (e.g. a Home Location Register for UPT services)). From the results shown in chapter 5 for the OAM error monitor, we see that a cell error rate of 4e-03 will result in a mean link changeover time of between 10 and 100 seconds (depending on the utilisation) for both the 1 Mbps and 5 Mbps links. We now examine the likelihood of these links, with this error rate, being returned to service by the ITU link alignment procedure.

We shall assume that the ITU link alignment shall operate continuously for 10 minutes (the upper limit for the SS7 case) before retiring a link. Each (unsuccessful) proving attempt takes T_p seconds, i.e. the time to establish an SSCOP connection, send 1000 cells, release the connection, then wait 5 seconds (i.e the delay before the next proving attempt). If we assume that the time to establish/release an SSCOP connection is 100 msec, then T_p is 5.37 seconds for the 5 Mbps link and 6.05 seconds for the 1 Mbps link. The number of proving attempts (N_p) in 10 minutes is then 111 and 99 for the 5 Mbps and 1 Mbps links respectively.

The probability P_f of failing a proving attempt is simply the probability of at least 1 errored cell in 1000. For the example we are considering here, P_f is 0.9818. Hence the probability P_r that the ITU link alignment procedure will return the link to service within 10 minutes (only to be removed by the error monitor) is $P_r = 1 - P_f^{N_p}$. This is 0.87 for the 5 Mbps link and 0.84 for the 1 Mbps link. A similar calculation for a 64 kbps link gives a P_r of 0.45.

Hence we see a similar result to that reported in [RAM93] for SS7 links, i.e an error rate which will cause a link to be failed, then returned to service (with high probability) by the link alignment procedure. As the link in our example should not in fact be operating (according to the criteria established in chapter 5), then clearly the ITU link alignment procedure needs adjustment (rather than the OAM error monitor). This example assumes however that link alignment attempts continue for 10 minutes (i.e. the SS7 procedure). As indicated in [RAM93], it may be desirable to reduce this time, thereby providing fewer proving attempts and hence a smaller probability of re-admitting a bad link.

We now examine the performance of the ITU proving scheme when errors are bursty. In particular we consider the bursty error example from chapter 5, which has alternating high and low error periods with exponentially distributed lengths. To simplify our example, we shall assume a BER of zero during the low error periods,. The BER during the high error periods remains at 3e-04, as in chapter 5. If we assume the high and low error periods to have mean lengths of $T_{high} = 50$ msec and $T_{low} = 250$ msec respectively, then the resulting mean cell error rate is 2e-02. As shown in Figure 5.4a and Figure 5.4b, the OAM error monitor will remove both the 1 Mbps and 5 Mbps links from service in less than 10 seconds under these circumstances. We now determine the probability that the ITU link alignment procedure will return the links to service.

The probability P_s that a proving period is successful for the bursty error case is

$$P_S = p\,(proving\ starts\ in\ low\ error\ period) \qquad \text{(Eqn 6.1)}$$
$$p\,(low\ error\ period > T_{1000})$$
$$= (T_{low}/(T_{high} + T_{low}))\, e^{-T_{1000}/T_{low}}$$

where T_{1000} is the time for 1000 cells to arrive. Equation 6.1 assumes that proving attempts which contain any portion of a high error period will fail. This ignores the possibility of a proving period which contains a high error period which does not actually produce any errors (and hence results in successful link proving). The value obtained here for P_S is therefore conservative (i.e. less than its actual value). Similarly, the value for P_r, the link restoration probability ($= 1 - (1 - P_S)^{N_p}$) is also less than its actual value. This conservative estimate for P_r is not of great concern in the context of this thesis, as our aim is to investigate whether the current ITU link proving scheme can return a bad link into service under bursty error conditions (in other words, whether P_r is large when errors are bursty).

Applying Equation 6.1, we get P_f values (i.e. $1 - P_s$) of 0.97 and 0.58 for the 1 Mbps and 5 Mbps links respectively. The corresponding probabilities P_r that the ITU link alignment procedure will return the links to service in 10 minutes are 0.95 and (essentially) 1.0 for the 1 Mbps and 5 Mbps links respectively.

A similar calculation for the 64 kbps link however results in a probability of (essentially) 0 that the link will be returned to service. This is due to the longer period for 1000 cells to pass (13.25 seconds, as opposed to 848 msec and 170 msec for the 1 Mbps and 5 Mbps links respectively). This longer period is sufficient to detect bursty error situations like those considered in the example. Clearly then, using a fixed number of cells for proving means that the proving procedure detects bursty error conditions less effectively as the peak rate increases.

It would appear that the ITU proving procedure is not suitable for the links considered here, as it returns poorly performing links to service (resulting in link oscillations). This is despite the claim in Q.2140 that the proving procedure will "provide satisfactory performance over a wider range of operating environments" than the 64 kbps links for which it was designed. The problems seen here with the ITU proving scheme are due to two factors:

1) The time taken for a proving attempt is too short for high speed links (i.e. 848 msec and 170 msec for the 1 Mbps and 5 Mbps links respectively). As shown in the example, these short proving intervals mean a high likelihood that bursty error conditions will go undetected.

2) Too many proving attempts are allowed (note however that Q.2140 does not specify that link alignment is to continue for 10 minutes. We have assumed this from the SS7 procedure described in [RAM93]).

Clearly a proving period of less than 1 second, the case for the two links here, is insufficient. Hence at the very least, the ITU parameters should be scaled to better suit these higher peak rates. More generally though, the link proving scheme should be designed to meet a given performance objective over a specified range of link peak rates. Such a design is now presented.

6.5.5 A Better Link Proving Scheme

We begin by outlining the objectives of the ATM signalling link proving scheme. These are as follows:

1) The probability that a bad link (i.e. one which has more than the sustainable error rate) is re-admitted within the period of link alignment (e.g. 10 minutes) should be below a given bound (e.g. 1%).

2) A link which is performing adequately should be returned to service as quickly as possible, to act as a backup for the (new) primary link. This reduces the possibility of losing all signalling capability to a given destination, should the new primary link (which is carrying the load of the link under test) fail as well.

3) The proving scheme must work over a specified range of peak rates, ideally with a single set of parameter values.

Requirements 1 and 2 are at odds with each other, in the sense that while a longer proving period reduces the probability of re-admitting a bad link[5], it also increases the time before a good link is returned to service. Ideally, requirements 1 and 2 should be decoupled. To do this, we propose a provisional recovery scheme.

Provisional Recovery

The proving scheme in Q.2140 has two phases. The first is the SSCOP connection establishment which, if successful, indicates that the link can pass at least some traffic (SSCOP connection establishment is the sole proving requirement in our minimal MTP2, as well as being the emergency proving procedure in Q.2140). The second phase measures the error rate, by requiring a specified number of cells (i.e. 1000) to pass without error.

These two phases can be associated with two types of link failure. The first type is a complete link failure (i.e. cable cut), which may be recovered by a protection switch (taking about 50 msec [SEX92]) or by network restoration (taking about 2 seconds [GRO91]). The SSCOP connection establishment is an adequate test for the recovery of this first type of failure (i.e. if SSCOP connection establishment messages can be exchanged, then the link is back up). The second failure type is an due to impaired transmission. Here a

5. The length of the proving period (i.e. the number of cells) is different from the amount of time during which link alignment occurs (which we have taken to be 10 minutes). Increasing the latter effectively increases the number of proving attempts, thereby increasing the probability of re-admitting a bad link.

longer test, which measures an error rate, is needed (i.e. the second phase of the Q.2140 proving scheme).

Hence a previously failed link on which an SSCOP connection has been re-established can provide at least some service (and will be completely ready if the failure was of the first type, i.e. a cable cut). We propose then that links in this state (i.e. SSCOP connection up, but yet to pass phase 2 of the proving procedure) should be marked as "provisionally" recovered. A provisionally recovered link will only enter service if there is no other link to a given destination. This two stage recovery process reduces the period during which only a single link is available, thereby reducing the need for a quick (and possibly insufficient) second phase of link proving. This achieves the desired decoupling of the two requirements outlined above. When phase 2 of the proving completes, a changeback to the original link is done.

Further details of the provisional recovery scheme are as follows:

- As the provisional backup link is not fully tested, it may fail quickly if it goes into service (i.e. should another changeover occur). In the event of a chronic impairment in both links to a given destination, the provisional recovery scheme described here would cause changeovers/changebacks to continue indefinitely. To avoid this, a link would be retired from service if it failed more than a given number of times (e.g. 4) within a set period (e.g. 30 minutes).

- As a bearer restoration[6] (e.g. the recovery of a cable cut by reallocating network capacity [GRO91]) may take up to 2 seconds, the initial attempt to establish the SSCOP connection should last for longer than this. This may be achieved by setting two SSCOP variables: Timer_CC (which determines interval between the transmission of SSCOP connection establishment (i.e. BGN) PDUs) and MaxCC (which determines the number of BGN PDUs

6. We refer to bearer restoration as the return to service of a previously disrupted transmission facility. Signalling link recovery, by contrast, occurs when MTP2 (or its ATM equivalent) indicates that a previously failed signalling link may once more carry traffic.

send before the connection establishment attempt is aborted). For example, a value of 100 msec for Timer_CC and 30 for MaxCC would result in 3 seconds of SSCOP connection establishment attempts.

- If the connection establishment is not successful after 3 seconds, indicating an on-going link (or ATM switch) failure, then another connection establishment attempt should occur after a short interval (e.g. 5 seconds). These SSCOP connection establishment attempts would continue until the link alignment process was halted (e.g. after 10 minutes).

- Once an SSCOP connection has been established, it should remain in place, even if the subsequent proving attempts fail (Q.2140 releases the SSCOP connection after each failed proving attempt).

The net result of the provisional recovery scheme is to reduce the possibility of losing all signalling capability between two SPs. By reducing the coupling between the requirement for quick signalling link recovery and adequate link proving, the link proving period may be extended (if desired). We now outline a proposal for determining whether the error rate on an ATM signalling link is sufficiently low for it to return to service (i.e. phase two of the proving procedure)

Measuring Error Rate

The previous section showed that using a fixed number of cells for link proving resulted in unsatisfactory performance (i.e. inconclusive link proving) as the peak rate increased, due to the inability to detect bursty error conditions. To address this, we propose a proving scheme which requires a satisfactory error performance for a specified time (rather than over a set number of cells). To provide a context for this proposed proving scheme, we assume the following:

- Signalling peak rates ranging from 1 Mbps to 5 Mbps (these represent the upper and lower values for the signalling links considered in chapter 5).

- Single cell proving PDUs used, as in Q.2140.

- The utilisation of links undergoing proving is 0.1 (rather than 0.5 as in Q.2140). The reasons for this lower utilisation were outlined in the previous subsection.

- Links undergoing proving must show a mean cell error rate of less than 1/1000, as in Q.2140. This complements the maximum sustainable cell error rates obtained in chapter 5 (all of which exceed 1/1000 by about a factor of 4, or more).

- Link alignment attempts (i.e. continuous link proving) continue for 10 minutes after a link failure (the case in SS7).

We shall design the proving scheme so that a link with an error probability of 4e-03 (i.e. the worst case figure used for the example in the previous sub-section) will have less than a 1% probability of returning to service during this time. Similarly we shall specify, for the sake of this example, that the bursty error condition used in the previous example (alternating exponentially distributed high and low error periods, with means of 50 msec and 250 msec respectively, and with a cell error rate of 2e-02 during the high period) must also result in a probability of less than 1% of returning the link to service over the 10 minute interval.

The proposed proving scheme requires all links to display an error rate of less than 1/1000 for 20 seconds, regardless of the peak rate. This interval is chosen so that the link with the lowest peak rate (i.e. 1 Mbps), operating at a utilisation of 0.1, will send over 4000 cells. This provides a more conclusive test for an error rate of less than 1/1000 than the Q.2140 sample size (i.e. 1000 cells), and, as will be shown, meets the required performance objective (less than 1% probability of re-admitting a bad link).

The number of bad (i.e. errored cells) allowed during the 20 seconds will depend on the peak rate (e.g. no more than 4 bad cells allowed for the 1

Mbps link, no more than 8 for a 2 Mbps link and so on). To avoid multiple parameters, to match multiple peak rates, it is proposed that the proving algorithm simply counts the number of cells sent, N_C, and the number of errored cells detected N_e[7] (note that each cell constitutes an SSCOP PDU. Hence, due to the SSCOP sequence numbering scheme, errors due to missing cells are also detected). If the resulting cell error rate, N_C/N_e, exceeds 1/1000, then proving period is considered to have failed, and another one begins immediately. Conversely, a measured cell error rate of less than 1/1000 indicates a successful proving period.

If we define the maximum number of errors allowed in a proving period as $N_{max} = int(N_C/1000)$, then the probability of a successful proving period is

$$P_S = p(0\,errors) + p(1\,error) + \ldots + p(N_{max}\,errors) \qquad \text{(Eqn 6.2)}$$

$$= \sum_{i=0}^{N_{max}} b(i, N_C, p_c)$$

$$\approx \sum_{i=0}^{N_{max}} \frac{(N_C p_c)^i e^{-N_C p_c}}{i!}$$

where p_c is the cell error rate. This expression uses the poisson approximation for the binomial.

The actual count of testing PDUs sent, and errors detected, would be maintained by the Signalling AAL Layer management (methods for reporting to Layer Management the number of PDUs sent, and the number of errors, were outlined in chapter 5, section 4) We therefore propose that, while the ATM error monitor should use OAM cell blocks (for the reasons outlined in chapter 5), the link proving scheme (which complements the error monitor) should instead use SSCOP PDUs. The reason for using SSCOP PDUs in the latter case is that all proving PDUs are the same size (one cell). The prob-

7. This method of determining the cell error rate, namely counting the number of arriving cells, the number of errors, and taking the quotient, was suggested by one of the thesis examiners.

lems caused by the variation in the number of cells per MSU is the reason that we have not located ATM error monitor within the signalling AAL as well.

Applying Equation 6.1, and replacing T_{1000} with 20 seconds, shows that the probability of returning either the 1 Mbps or the 5 Mbps links to service during the bursty error period is essentially 0 (thereby satisfying the proving scheme objective). This is due simply to the increased proving time: 20 seconds, as opposed to less than 1 second with Q.2140.

We now examine whether this proposed proving scheme meets its stated objective (a less than 1% chance of returning a bad link to service) when errors are non bursty. Essentially the problem is to determine, for a given cell error rate p_c, the probability P_r that the proving requirements will be met for a 20 second period within a 10 minute window (in other words, when the number of consecutive proving attempts, N_p, is 30) Applying Equation 6.2 to find P_S, we get $P_r = 1 - (1 - P_S)^{Np}$. For the cell error probability considered in our example, 4e-03, we get $P_r = 0.13\%$ for the 1 Mbps link and (essentially) 0 for the 5 Mbps link, thereby meeting the 1% objective.

This subsection has proposed an improved ATM link proving algorithm, which has been systematically designed to meet given performance objectives. This algorithm has the following features:

- A provisional recovery facility is provided, which allows links which are under testing to be returned to service if required. This would be used only if the alternative was a complete loss of signalling capability.

- A low (and guaranteed) probability of returning a bad link to service, which operates over a range of peak rates. This is contrasted with the proving scheme in Q.2140, which while claiming to operate effectively over all peak rates, in fact does not.

6.6 Conclusions

This chapter concludes the investigation of the B-ISDN MTP presented in this thesis. Chapters 4 and 5 focused on ATM signalling link performance modelling and error monitoring. This chapter incorporates these results into a wider investigation of the protocol functions required for the B-ISDN MTP. The results and recommendations arising from this investigation are gathered here.

Central to the issue of B-ISDN MTP protocol functions is the signalling architecture which they will support. While many authors have proposed associated mode (i.e. fully meshed) B-ISDN signalling network architectures, there have been no studies of the resulting bandwidth requirements or call handling capacities (i.e. compared to STP based alternatives). Accordingly we have conducted a case study of a regional ATM signalling network which compares these two architecture types. The findings are as follows

- The bandwidth requirements of a fully meshed signalling network, while far greater (i.e. more than an order of magnitude) than those for an STP based network, are only a small proportion of the overall network bandwidth. For our study the figure was around 5%.

- This fully meshed bandwidth result was obtained using the simplest bandwidth allocation method, i.e. peak rate allocation. While statistical multiplexing the signalling links in a fully meshed network will result in bandwidth savings, other problems arise, in particular potential performance degradation due to cell loss in ATM multiplexers. We have not considered statistical multiplexing for ATM signalling in this thesis. However given that peak rate allocation (i.e. a worst case) still uses only a small fraction of the network bandwidth, it presents a viable (and simple) means for operating a fully meshed ATM signalling network.

- The fully meshed network call capacity is greater than the STP based alternative when a) all SPs generate calls, with destinations spread evenly throughout the network or b) all SPs generate calls to a single destination. This latter scenario has been identified as being of concern for SS7 networks [RUM94.1]. The reason is that any STP overload (at MTP3) resulting from this calling pattern degrades the entire signalling network performance, as MSUs from all calls use MTP3. By contrast, we observe that the fully meshed network restricts the effects of such an overload to the signalling links going to the target destination. This localisation of the effects of congestion is a major advantage of the fully meshed approach.

The main conclusion arising from the case study is that, in terms of bandwidth requirements and call handling capacity, the fully meshed approach is a viable alternative to using STPs. If fully meshed networks (or more generally, networks employing associated mode signalling) are used, then their particular MTP protocol requirements must be assessed. Accordingly the remainder of the chapter has examined the B-ISDN MTP protocol requirements for associated mode signalling. Our major findings are outlined below.

As private ATM networks are likely to have less stringent signalling reliability requirements than their public counterparts, a private ATM network may use a simpler MTP than the one proposed by ITU recommendation Q.2140 and [ATM94.1] and [ATM94.2]. Accordingly this chapter has proposed a minimal (and hence low cost) MTP. Its major features are:

- the use of the UNI Service Specific Coordination Function (SSCF) rather than the more complex NNI SSCF proposed in Q.2140.

- the use of SSCOP capabilities (such as link failure detection) to replace the equivalent functions in Q.2140.

- the elimination of all MTP3 management messages.

This proposed "minimal" MTP provides the basic functions of message transfer, link changeover and congestion control for private networks using associated mode signalling. The more fully featured (and hence more complex) MTP outlined in Q.2140, [ATM94.1] and [ATM94.2] is also oriented to the needs of associated mode signalling. However the additional maintenance functions, and the more sophisticated error monitoring/link changeover functions of this second MTP proposal are better suited to the needs of public (rather than private) networks. In particular, this second MTP would use the ATM error monitor proposed in chapter 5.

To complement our work on ATM error monitors, we have examined the ATM link proving scheme proposed in Q.2140. This is found to have two serious shortcomings. Firstly it precludes the use of statistical multiplexing for signalling traffic (while we have not proposed the use of statistical multiplexing, it appears unwise to implement a protocol which prevents this option). Accordingly we have outlined modifications to the proving scheme to allow the statistical multiplexing option.

The second shortcoming of the link proving scheme of Q.2140 is that it does not detect poorly performing links. The aim of link proving to determine if a signalling link is essentially error free (i.e. the error rate is less than a specified level). However the scheme in Q.2140 is not an adequate test in this regard, particularly if bursty errors prevail. We have proposed an alternate proving scheme, which requires a signalling link to demonstrate the required error for a given period of time, rather than for a given number of cells (the approach in Q.2140). The revised link proving scheme we propose has, as an added advantage, the ability to operate over a range of signalling peak rates.

7. Conclusions

7.1 Overview

B-ISDN will have a major impact on signalling transport. This is due to the use of ATM VCCs to carry signalling traffic, instead of SS7 links. The fundamental differences between ATM VCCs and SS7 links mean that the B-ISDN Message Transfer Part (MTP) will be substantially different from its SS7 counterpart.

While the B-ISDN signalling standards relating to the MTP have developed considerably over the last few years, many important issues remain unresolved. In particular, the following areas require investigation:

- delay performance modelling of ATM signalling links

- ATM signalling error monitoring

- B-ISDN signalling network architecture

- B-ISDN MTP protocol functions

This thesis has concentrated on these four aspects of the B-ISDN signalling MTP. Our results and recommendations are described in the following sections, as well as further areas of research.

7.2 Performance Modelling

The ITU has introduced a new link level protocol, known as the Service Specific Connection Oriented Protocol (SSCOP), which will be used for ATM signalling. As the SSCOP differs substantially from its SS7 counterpart, due to the use of Selective Repeat error correction instead of go-back-n, a new delay performance model is needed. Such a model has been presented in this thesis. In addition to incorporating Selective Repeat operation, and the error reporting procedures of the SSCOP protocol, this model has the following features:

- The increased queueing delay, due to a mixture of different Message Signal Unit lengths, is incorporated (rather than assuming all MSUs to be the same size).

- The effect of the change in error probabilities for MSUs of different lengths (within a given MSU length distribution) is accounted for (rather than assuming the MSU error probability to be constant).

- Delay percentiles are provided, in addition to the mean delay.

- Variations in round trip delay (in particular between the first retransmission of a failed packet and a subsequent retransmission of that packet) are modelled.

As well as giving accurate SSCOP delay results, this model provides a general result for Selective Repeat delay. This new Selective Repeat model overcomes deficiencies of earlier Selective Repeat analyses.

The SSCOP delay expressions developed in this thesis are used to determine maximum sustainable error rates for ATM signalling links. These maximum error rates form the basis of the ATM signalling error monitor design.

7.3 Error Monitoring

In addition to revising techniques for delay performance modelling, a new approach is needed for ATM error monitoring. This is due to the differences between SS7 links and ATM VCCs, in particular the variable bit rate on ATM signalling links (SS7 links have a constant bit rate), and the performance monitoring capabilities of the SSCOP and the ATM Operations And Maintenance (OAM) layers.

Three potential ATM error monitoring schemes have been examined in this thesis. These either:

- count errored Signal Units (like the SS7 Signal Unit Error Rate Monitor (SUERM)).

- count errored time intervals (like the SS7 Errored Interval Monitor (EIM)).

- monitor blocks of ATM cells, using the OAM capabilities.

The basis for this examination has been how well these scheme fulfil the following three requirements, i.e. that they:

1) cause a link changeover as soon as possible after the maximum error rate is exceeded.

2) tolerate burst errors for a specified period before initiating changeover.

3) ensure that the changeover transient (i.e. the amount of data queued when changeover is initiated) does not exceed a specified value.

As a result of this investigation, we have made a number of observations regarding these three error monitoring schemes. These are as follows.

The SUERM (also known as the Leaky Bucket) has two major shortcomings in an ATM environment. Due to the variability in MSU size, it is not possible to dimension the SUERM so that it limits the changeover transient while also guaranteeing a minimum changeover time. We have proposed a method for overcoming this first problem, namely a separate timer which ensures a minimum changeover time. However the SUERM also features a much longer changeover time than the EIM or OAM monitors on high speed links, due to the limitations imposed by the MSU size.

Similarly the Errored Interval Monitor (EIM), proposed for high speed SS7 links, has two serious shortcomings in an ATM environment. The first is that the interval error probability changes with the link utilisation (due to the change in the number of cells per interval). This results in conservative changeovers at high utilisations. The second shortcoming is that, due to the way the Signalling ATM Adaptation Layer (AAL) detects and reports errors, there is likely to be a lower limit on the EIM interval size. This lower limit may restrict the ability of the EIM to detect bursty error situations in an ATM environment.

Our conclusion is that the OAM scheme is the most suitable ATM error monitor. However, to be able to detect bursty errors, as well as provide a rapid changeover when non bursty errors prevail, the OAM error monitor needs to operate with two block sizes. A easily implemented algorithm has been designed which allows this.

7.4 Network Architecture

Due to the low cost of ATM signalling VCCs, compared to SS7 links, a fully meshed ATM signalling architecture may be feasible (as opposed to a network which uses Signal Transfer Points (STPs)). However the increased bandwidth requirements of the fully meshed approach require investigation, as well as the respective call handling capacities of the two signalling architectures. A comparative study of these two potential ATM signalling architectures (i.e. fully meshed and STP based) has been presented which considers these two aspects (such a study has not been done previously). The conclusion is that the fully meshed approach should not be precluded on the basis of its bandwidth requirements. In addition, the fully meshed approach provides better delay performance and call handling capacity in many instances.

7.5 B-ISDN MTP

Assuming that fully meshed ATM signalling networks may be widely used, their MTP protocol requirements must be examined. A major task of the SS7 MTP is to ensure the integrity of STP based networks. However if STPs are not deployed, a simplified MTP can be used.

We have examined closely the protocol functions needed for this simplified B-ISDN MTP. Resulting from this examination is a B-ISDN layer 3 MTP (i.e. MTP3) proposal. This provides the minimum capabilities needed to transfer MSUs in a private network which uses B-ISDN Capability Set 1 (CS-1) signalling only. The aim of this minimal MTP3 is to allow a low cost implementation. A more complex B-ISDN MTP3, which provides the failure recovery and maintenance features required for public networks, has been proposed elsewhere [ATM94.1] [ATM94.2].

To complete the B-ISDN MTP work, the MTP2 functions required for B-ISDN have been considered. In line with the proposed minimal MTP3, a simplified B-ISDN MTP2 has been described. This is contrasted with the fully featured B-ISDN MTP2, i.e. the one outlined by the ITU [Q.2140]. To complement our work on ATM error monitoring, we have examined the link proving scheme proposed in [Q.2140] (which determines when a failed ATM Signalling link may be returned to service). This is found to be deficient in a number of regards. In particular there is, for some peak rates, a high probability of re-admitting a bad link into service when bursty errors prevail. Accordingly we have proposed a new ATM signalling link proving scheme, which places an upper bound on the probability of re-admitting a bad link. This new proving scheme is designed to operate over a range of signalling peak rates.

7.6 Future Work

B-ISDN signalling transport is a new area. As B-ISDN systems are only now being implemented, there is not the background of previous experience which, for example, guides SS7 implementations. While this thesis has examined important aspects regarding the implementation and engineering of the B-ISDN MTP, there are areas requiring further research. These areas are outlined below.

Statistical Multiplexing

The thesis has shown that peak rate allocation is feasible for fully meshed regional B-ISDN signalling networks (e.g. comprising around 40 nodes). However there are proposals for fully meshed B-ISDN networks [GID93], which are an order of magnitude larger. Such networks, if deployed, will most probably require signalling traffic to be statistically multiplexed.

This thesis has not investigated statistical multiplexing of signalling traffic. Such an investigation would need to consider bandwidth requirements, and responses to congestion (i.e. cell loss) in the ATM layer. These two aspects are general problems to be considered for all statistically multiplexed ATM services. As such, they are active research areas (e.g. [HUG91] [BIA91] for bandwidth requirements, [ATM94.3] [ATM94.4] for congestion control). However a particular concern for statistically multiplexed signalling traffic is the interaction between ATM layer congestion controls (if used) and error monitors when cell loss (due to buffer overflow occurs). In particular, the error monitor would need to differentiate between cell loss due to a deteriorating link (when a changeover would be needed) and cell loss due to ATM layer congestion (when a changeover could potentially be disruptive).

Dynamic Bandwidth Re-Allocation

Capability Set 2 signalling will allow VCC bandwidth to be changed dynamically during a call. This will make possible a different approach to link level congestion control, i.e. instead of reducing the load on a congested signalling VCC, its bandwidth may be increased temporarily. This is potentially an attractive option, as the cost of the extra signalling bandwidth could be more than offset by the revenue gained by admitting extra calls (which may be lost if standard congestion control procedures are applied). There are many issues which would need to be addressed if this approach were used: for example resolving contention when many signalling VCCs request bandwidth simultaneously (i.e. during a focused overload) and determining parameters such as the amount of capacity increase and when to revert to the former capacity level.

B-ISDN Signalling Arrival Models and Error models

As with previous signalling delay analyses, we have assumed poisson arrivals and independent bit errors (however unlike previous analyses, we account for the change in error rate with MSU size. In addition we consider bursty errors in the context of error monitoring and link proving schemes). There is scope for further delay performance analyses, where these assumptions are relaxed.

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Appendix A. Paper: Leaky Bucket and M/G/1/K queue modelling

A.1 Introduction

This appendix contains a paper describing the Leaky Bucket modelling procedure introduced in chapter 3. The paper extends this procedure to give the mean first passage time (i.e. mean time until the first overflow) of the M/G/1/K queue.

First Overflow of the M/G/1/K queue, with application to Leaky Buckets in SS7 signalling

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A method is shown for determining the mean and variance of the number of arrivals until the first overflow of an M/G/1/K queue, given the initial queue length. Continuous and discrete time cases are considered. The method is applied to the SS7 Leaky Bucket.

1 Introduction

Although M/G/1/K queueing systems have been widely studied [1,2], most analyses have concentrated on the steady state distribution of the queue length. The LST of the time for the first overflow of an M/G/1/K queue is given in [4], however the expression becomes intractable for large values of K. We use an alternative method to find the mean and variance of the number of arrivals until the first overflow, given the initial queue length. This has application in communication systems which use Leaky Bucket techniques, such as SS7 signalling [3].

2 Analysis

Initially we follow the approach for the steady state M/G/1/K queue [1]. We obtain the transition matrix for the imbedded Markov Chain formed by the queue length immediately after departures. This is a K x K+1 matrix with the form

$$P = \begin{bmatrix} p_0 & p_1 & p_2 & \cdot & \cdot & \cdot & p_{K-1} & 1-\sum_{i=0}^{K-1} p_i \\ p_0 & p_1 & p_2 & \cdot & \cdot & \cdot & p_{K-1} & 1-\sum_{i=0}^{K-1} p_i \\ 0 & p_0 & p_1 & \cdot & \cdot & \cdot & p_{K-2} & 1-\sum_{i=0}^{K-2} p_i \\ 0 & 0 & p_0 & \cdot & \cdot & \cdot & p_{K-3} & 1-\sum_{i=0}^{K-3} p_i \\ \cdot & \cdot & \cdot & \cdot & \cdot & \cdot & \cdot & \cdot \\ 0 & 0 & 0 & \cdot & \cdot & p_0 & p_1 & 1-p_0-p_1 \end{bmatrix} \tag{1}$$

where

$$p_n = \int_0^\infty e^{-\lambda t}(\lambda t)^n/n!\,dB(t) \tag{2}$$

$B(t)$ is the service time distribution function and λ the mean arrival rate. The second to last column gives the probabilities of the queue filling (but not overflowing) during a service interval.

The imbedded Markov Chain described by this matrix has $K-1$ transient states (corresponding to queue lengths $0, 1, ..., K-1$), and one absorbing state. This is entered the first time the queue overflows. The state associated with a queue length of K is not included, as it can only be reached from the absorbing state, given that we consider the queue length immediately after departures.

The theory of absorbing Markov Chains [5] is now applied. A sub-matrix Q is formed, which contains the transition probabilities of the transient states. This is just $\hat{P}$ without the final column. We now define a matrix N where

$$N = (I - Q)^{-1} \tag{3}$$

and I is the identity matrix. From [5] we now obtain vectors containing the mean and variance of the number of visits to the transient states (e.g. queue lengths 0,1,..,K-1) until the first overflow, given an initial queue length i ($0 \leq i \leq K-1$). These are

$$M = Ne \tag{4}$$

for the mean and

$$V = (2N - I)M - M_{sq} \tag{5}$$

for the variance. M_{sq} is M with each element squared and e is a column vector of ones. As each visit to a transient state corresponds to a customer being served, we now have the mean and variance of the number of customers served until the first overflow, given an initial queue length. The mean number of arrivals (which we seek) is the number of customers served plus $K+1$.

This result is extended to a discrete time case, where p_j = probability[service time = j slots] and p is the probability that a customer arrives during a slot. The transition probabilities become

$$p_n = \sum_{j=n}^{\infty} p \binom{j}{n} p^n (1-p)^{j-n} \tag{6}$$

3 Leaky Bucket

The Leaky Bucket algorithm has many applications in communication systems, one of these being error monitoring for SS7 signalling links. Here a "Leaky Bucket" counter is incremented for each Message Signalling Unit (MSU) in error. In addition, the counter is decremented (provided that it is not already zero) for every D Signalling Units. A faulty link is declared when the Leaky Bucket counter reaches a level T, with the current CCITT values for T and D being 64 and 256 respectively. We wish to know the mean number of Message Signalling Units required to overflow the Leaky Bucket, given an MSU error probability p.

We model the Leaky Bucket as a discrete time queue, where MSU errors are treated as customer arrivals. The queue overflows at the level T, so that the number of spaces in the queue (K in our previous notation) is T-1. A customer is served (e.g. the counter is decremented) every D slots, where each MSU arrival is one slot. The probability of an customer arrival (e.g. an MSU error) in a slot is the error probability p. The transition probabilities are then

$$p_n = \begin{cases} \binom{D}{n} p^n (1-p)^{D-n} & 0 \le n \le D \\ 0 & D < n \end{cases} \tag{7}$$

The mean number of errors until the Leaky Bucket overflows (assuming that it starts empty) is

$$M(1) + T \tag{8}$$

and hence the mean number of MSU arrivals until overflow is $(M(1) + T)/p$.

Figure 1 shows the mean time (in minutes) to overflow a Leaky Bucket with the CCITT parameters (T = 64, D = 256), assuming an MSU arrival rate of 200/sec. The MSU error probability is in units of 1/D. Similar results (using a different method) are given in [6].

4 Conclusion

A method has been presented for determining the mean and variance of the number of arrivals before the first overflow of an M/G/1/K queue, given the initial queue length. Both continuous and discrete time cases are considered. The method is used to find the mean number of packets to overflow an SS7 Leaky Bucket counter.

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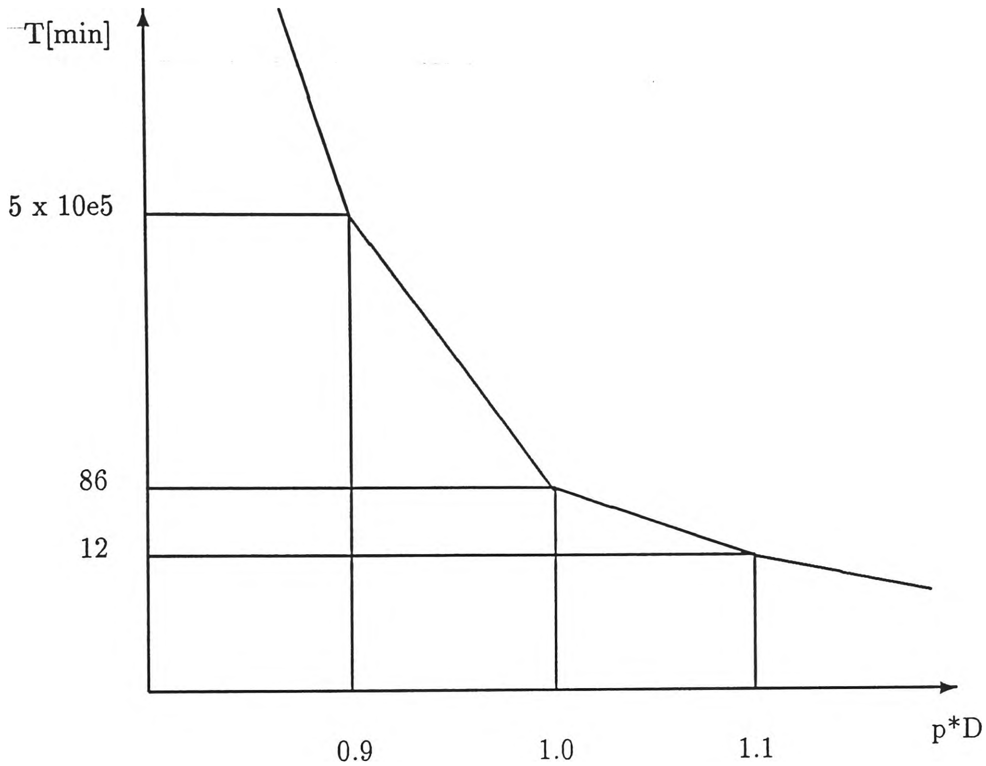

Figure 1: Overflow time versus relative error rate

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